Aastra 6700i SIP Terminals for MX-ONE

INSTALLATION INSTRUCTIONS

AASTRA

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1 General

This document is valid for Aastra 6730i, 6731i, 6735i, 6737i, 6739i, 6753i, 6755i and 6757i SIP phones, when installing these telephones in a MX-ONE environment.

1.1 Scope

This document describes how to install and configure for the Aastra 6700i terminals in a MX-ONE Telephony Server (TSE) environment. For general installation information that is not unique for a MX-ONE environment, there is a reference to the Administrator Guide for Aastra Models 6700i and 9000i Series IP SIP Phones.

There is also one platform independent installation guide per telephone model available on www.aastra.com.

1.2 Glossary

Some expressions in this document follows the expressions used in MX-ONE, which can differ from the expressions used in the Administrator Guide for Aastra Models 6700i and 9000i Series IP SIP Phones.

Table 1 Expressions used in MX-ONE and in 6700i documents

MX-ONE	Administrator Guide for Aastra Models 6700i and 9000i Series IP SIP Phones
Software server	Configuration server
Shortcut keys	Programmable keys and Softkeys
TNS (Telephony Name Selection)	Speed dial
MNS (Monitored extensions)	BLF (Busy Lamp Field)
Settings key	Options key
Key Panel Unit (KPU) and Display Panel Unit (DPU)	Expansion Module

1.3 Environmental Requirements

See Administrator Guide for Aastra Models 6700i and 9000i Series IP SIP Phones.

2 Cabling

See Administrator Guide for Aastra Models 6700i and 9000i Series IP SIP Phones.

3 Power Equipment

These telephones can be powered from any of the following methods:

- 6730i is powered from an AC adapter 5V. No PoE support.
- 6731i, 6735i, 6737i, 6739i, 6753i, 6755i, 6757i are powered using PoE according to IEEE 802.3af or from an AC/DC adapter 48 V.
- Power over Ethernet power injector, which supplies 48 V power through the Ethernet cable on pins 4&5 and 7&8. The power injector is available as an optional accessory.

Table 2 Power classes for the different phone models

Device	Power Consumption **)	Power Class
6730i	2.4 W	not applicable
6731i	2.4 W	1
6735i	2.8 W	2 *) from REV:29
6737i	2.9 W	2 *) from REV:30
6739i	4.8 W	0
6753i	3.3 W	0
6755i	4.0 W	0
6757i	4.1 W	0

*) Maximum one expansion module with PoE. Up to three modules with AC/AC adapter.

**) Single call established in handset mode, backlight on. No PC connected.

Explanation of power classes:

- 0 - classification is not implemented.
- 1 - less than 3.84 W.
- 2 - less than 6.49 W.

4 Earthing and Grounding

See Administrator Guide for Aastra Models 6700i and 9000i Series IP SIP Phones.

5 Setting up the Software Server

The software and the configuration files used by the IP phones shall be stored on a server where the IP phones can fetch them. The server is called IP Phone SW Server.

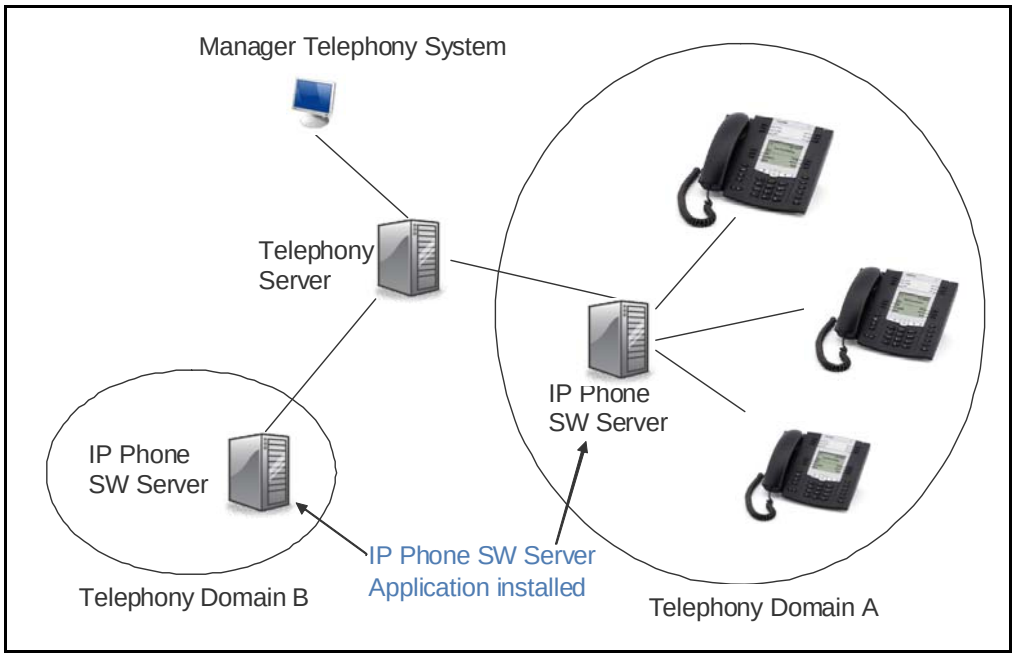


Figure 1: Deployment Scenario when telephony domains use different SW servers

In the Telephony Server you can define multiple telephony domains. The telephony domains are managed through the Manager Telephony System web interface.

The IP phone configuration files are preferably generated through the Manager Telephony System (MTS). To enable the files to be generated through Manager Telephony System, the *IP Phone SW Server Configuration Management Application* must be installed on the IP Phone SW Server. Once generated the configuration files can viewed directly on the IP Phone SW Server.

The IP phones can use the following protocols to download the software and configuration file(s): http, https, ftp, tftp. The recommendation is to use the http protocol and it is described in these installation instructions.

If MTS for some reason is not used, stop reading and go to chapter "How to start a new phone".

Setting up the software server comprises the following steps:

- Installing the IP Phone SW Server Configuration Management Application.

- Reconfigure Microsoft IIS web server, if it exists.
- Creating a directory structure on the IP Phone SW Server.
- Copy the IP phone application and language files to the IP Phone SW Server. The configuration files shall not be copied, these are created by MTS. For information about the files, see section 8.1 Phone software and configuration files on the software server on page 24.

5.1 Install IP Phone SW Server

Before the installation of the IP Phone SW Server you have to install the Java Runtime Enviroment. When you run IP Phone SW Server wizard the Tomcat is also installed.

Note: For more information see the IP Phone SW Server release notes for additional installation information. See also section 8.1 Phone software and configuration files on the software server on page 24 on how to manage the SW in the IP phone.

5.1.1 System and program requirements

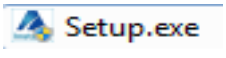
- Java Runtime Enviroment (JRE) version 6 (32- bit) or later.
- Windows 32 bit or 64 bit.
- Tomcat version 7.0.33 (apache-tomcat-7.033-windows x86.zip), is installed via the wizard.

5.1.2 Prerequisites

- Check if Java Runtime Enviroment version 6 or later, is installed. If not, installed JRE before you start the IP Phone SW Server wizard.
- If IP Phone Server (version 1.4 or prior) is installed, uninstalled the the program before you start the wizard.

5.1.3 Install IP Phone SW Server (and Tomcat)

Do as follows:

1. Download and click **Setup.exe**, 

. The window **Installer Language** is displayed.
2. Select **Language** from the list.

3. Click **OK**. The window **Welcome to the IPPhone SW Server Setup Wizard** is displayed.

IPPhone SW Server Setup
Welcome to the IPPhone SW Server Setup Wizard
This wizard will guide you through the installation of IPPhone SW Server.
It is recommended that you close all other applications before starting Setup. This will make it possible to update relevant system files without having to reboot your computer.
Click Next to continue.
Next >
Cancel

4. Click **Next**. The step **Licence Agreement** is displayed.
5. Click **Agree**. The step **Tomcat Port Number** is displayed.
6. Type the **port number** in the field, default port is 80.
7. Click **Next**. The step **Tomcat Administrator** is displayed.
8. Click **Next**.
9. The message window, **Do you want to continue without configuring a Tomcat administrator?** is displayed.
10. Click **Yes**. The step **Choose Install Location** is displayed.
11. Click **Install**. When the installation is complete you will get a message, **Installation Complete**.
12. Click **Next**. The step **Completing the IPPhone SW Server Setup Wizard** is displayed.
13. Click **Finish**.

5.2 Co-existence with Microsoft IIS Web Server

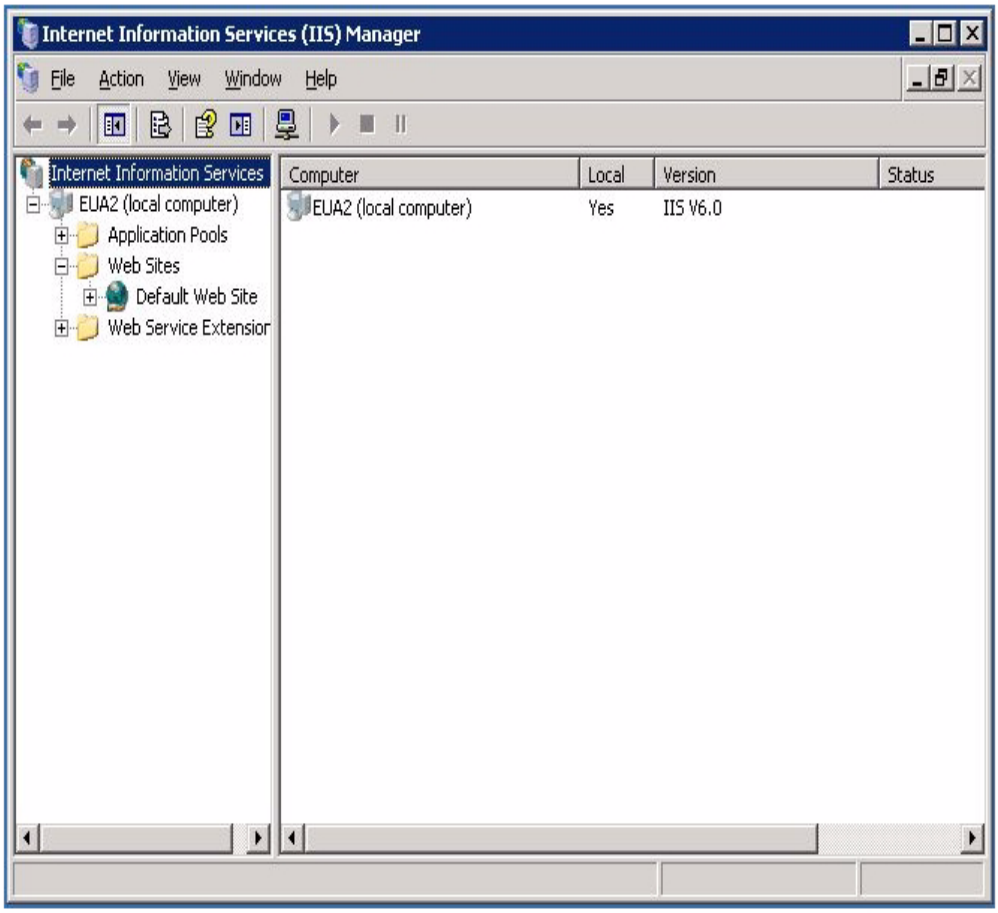
If a Windows IIS web server is running on the IP Phone SW Server there will be a port conflict with the IP Phone SW Server Configuration Management Application since they are both using port 80. This is typically the case when CMG or OneBox are installed on the same server.

The reason that you need to have the Tomcat web server running instead of just using the IIS web server is that the IP Phone SW Server Configuration Management Application is developed in Java and IIS can only host web applications developed in the Microsoft environment.

Perform the following steps to resolve the port conflict.

- Keep IIS running on port 80.
- Reconfigure the IP Phone Configuration Management Application to run on port 82 instead.
  - Edit the **jakarta-tomcat-4.1.31\conf\server.xml** (where -4.1.31 is an example) and change the port 80 to 82.
  - Restart Tomcat by going to `Control Panel/Administrative Tools/Services` Restart the service Jakarta Tomcat 4.
- Connect MTS to IP Phone SW Server Configuration Management Application on port 82, using the MTS task IP Phone SW Server.
- Create the configuration file in the MTS task IP Phone configuration File and it will be stores on the IP Phone SW Server, see section 6.1 Create a Configuration File on page 15 for details.
- The **.cfg**, **.st**, **.txt** and **.tuz** file types must be enabled. Follow the steps below to enable these file types:
  - In **IIS Manager**, select File Type, select **DefaultWEB Site**.
  - Select **Properties** and edit **HTTP header**. Apply the following settings:
  - **Associated extension: .cfg**, **.st, .txt and .tuz** (encrypted .cfg file)
  - **Content type (MIME): application/octet-stream**.
- Redirect IIS web server to Tomcat web server for the IP phone's requests like this:

– Open C:\WINDOWS\system32\inetsrv\inetmgr.exe, navigate to Default Web Site.



– Right click on Default Web Site and select New Virtual Directory. A wizard will start.

– Enter the directory name to where the telephone firmware shall be stored as Alias, example: aastra67xxi.

– Enter the path to the folder under Tomcat, example: C:\jakarta-tomcat-4.1.31\webapps\ROOT\aastra67xxi.

– Enable the Read option and finish the wizard.

– You can now access the Tomcat folder with terminal settings on both port 80 as well as 82, while MTS can update the configuration file on port 82.

– If subnets or telephony domains are defined for the configuration file in MTS, the path under Tomcat will include the subnet/telephony domain in its path. Update the IIS virtual directory link accordingly.

5.3 Start of the Tomcat Web Server

During the installation of IP Phone SW Server Configuration Management Application for Windows®, Tomcat will start automatically on port 80. In case Tomcat is not up and running, it can be started manually with Windows® Services Jakarta Tomcat 4 (see instruction below) or by restarting the Tomcat web server PC.

If the Tomcat web server starts on another port number than 80, the port number can be set manually in the file **jakarta-tomcat-4.1.31\conf\server.xml** (where -4.1.31 is an example).

5.3.1 Verify that the Apache Tomcat server is running

Use the following procedure on the server PC:

1. Go to `Control Panel/ Administrative Tools/ Services`

   Check that the service Jakarta Tomcat 4 has status **started**. If the service has not been automatically started, then start it manually.

2. Start a web browser for example Internet Explorer and enter the address `http://localhost`. The Apache Tomcat web page server is shown.

3. Enter the address `http://localhost/ipfmsBinary` in the web browser.

   The Axis web page is shown.

6 Manage the Configuration Files in Manager Telephony System

MX-ONE Manager Telephony System (MTS) shall be used when creating or changing the aastra.cfg and the model specific configuration files. The information regarding parameters is available in the online help for MX-ONE Manager Telephony System. The picture below shows an example of the page in the IP Phone Configuration File task in MTS:

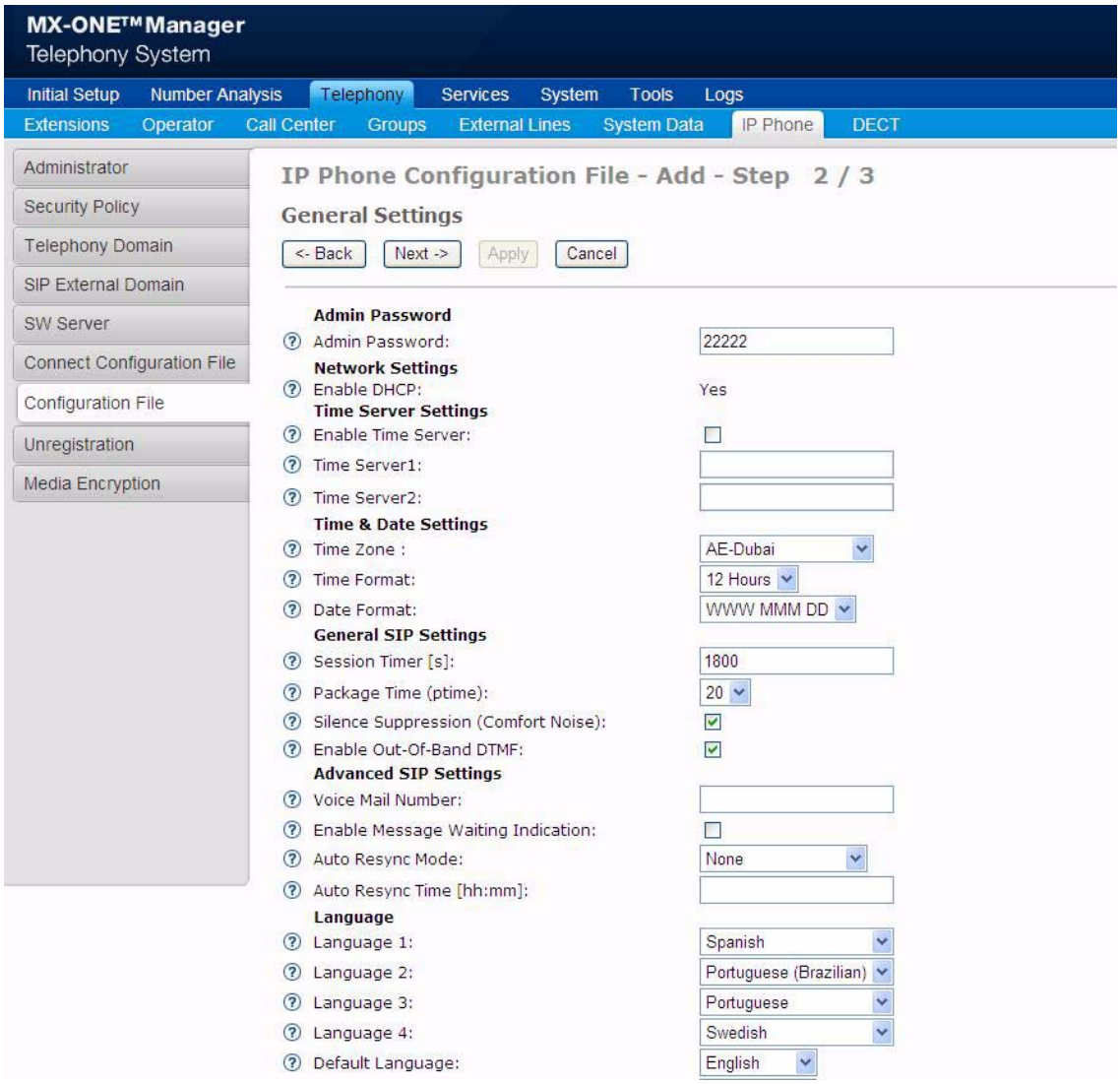


Figure 2: IP Phone Configuration File in Manager Telephony System

Note: MTS requires that the IP Phone SW Server Configuration Management Application is installed on the IP Phone SW Server, please see section 5 Setting up the Software Server on page 8

6.1 Create a Configuration File

The procedure to create a new configuration file is:

- Log in to MX-ONE Manager Telephony System.
- Go to **Telephony** > **IP Phone** > **Configuration file**. Press **Add new** to open the new configuration file.

  Make sure that **Aastra 67xxi** family is selected and enter the data into the configuration file which is automatically stored under the correct directory in the IP Phone Software Server when pressing **Apply**.
- To force the telephones to fetch the new configuration file there are a number of cases:
  - If the telephones are not started yet: connect the power and the telephones will fetch the new configuration file.
  - If the telephones are already registered to the PBX, select the **Unregistration** option to force the telephones to fetch the new configuration file.
  - The telephones will after less than 24 hours automatically fetch the new configuration file and if necessary download a new firmware.
  - Restart the telephones manually.

6.2 Changing an Existing Configuration File

The existing configuration file can be updated using the **MX-ONE Manager Telephony System**.

The following procedure shall be used when the configuration file shall be changed:

1. Log in to MX-ONE Manager Telephony System and select:

   Telephony > **IP Phone** > **Configuration file**
2. Take a backup copy of the existing configuration file by pressing the backup icon.
3. Use the **Change** icon to view the configuration file. When the adaptation of the file is completed, it is automatically stored under the **aastra67xxi** directory in the IP Phone Software Server.
4. For the telephones that are already registered to the PBX, select the **Unregistration** option to force the telephones to fetch the new

configuration file. For the not registered telephones, see section **6.1 Create a Configuration File on page 15**.

6.3 Scratch Pad when Creating the Configuration file

If a new parameter has to be added into the aastra.cfg file but there is no support for this new parameter in Manager Telephone System, the scratch pad can be used. Another usage is if MTS creates a parameter value, but another value is requested.

It is a free text window where the new parameter or parameter value can be entered. The parameters are added at the end of the **aastra.cfg** file. If a parameter exists twice in the configuration file, the telephone uses the value at the end of the file.

The scratch pad is found at the bottom of the page:

Telephony > **IP Phone** > **Configuration File** > **General Setting**

6.4 Connect Existing Configuration File to Manager Telephony System

In a system where MX-ONE Manager Telephony System has not previously been used when working with an IP phone configuration file, the existing configuration file can be connected to the MTS instead of having to be recreated. Follow the steps below to connect a configuration file to MX-ONE Manager Telephony System.

1. Log on to MX-ONE Manager Telephony System.
2. Go to **Telephony** > **IP Phone** > **SW Server** where you register the IP Phone SW Server.
3. Go to **Telephony** > **IP Phone** > **Connect Configuration File**.
4. Select the IP Phone SW Server and search for existing files. Click on the **Connect** icon next to the configuration file to connect to.
5. Go to the **Configuration File** task. Select the connected configuration file and use the **Change** icon to change the file, if needed. When the adaptation of the file is completed, it is automatically stored under the correct directory in the IP Phone Software Server.
6. For the phones that are already registered to the PBX, select the **Unregistration** option to force the phones to fetch the new configuration file.

6.5 Retrieve the Back-up Copy

If any problem is discovered when a new configuration file has been loaded into the phones and there is a need to go back to the previous version, the following procedure shall be used:

1. Log on to MX-ONE Manager Telephony System.
2. Go to **Telephony** > **IP Phone** > **Configuration File**.
3. Use the **back-up retrieve** icon.
4. For the phones that are already registered to the PBX, select the **Unregistration** option to force the phones to fetch the new configuration file. For phones that are not registered, 6.1 Create a Configuration File on page 15.

7 How to Start a New Phone

The phone is delivered with default settings for an IP network. These settings must be adapted to the local network using phone configuration files.

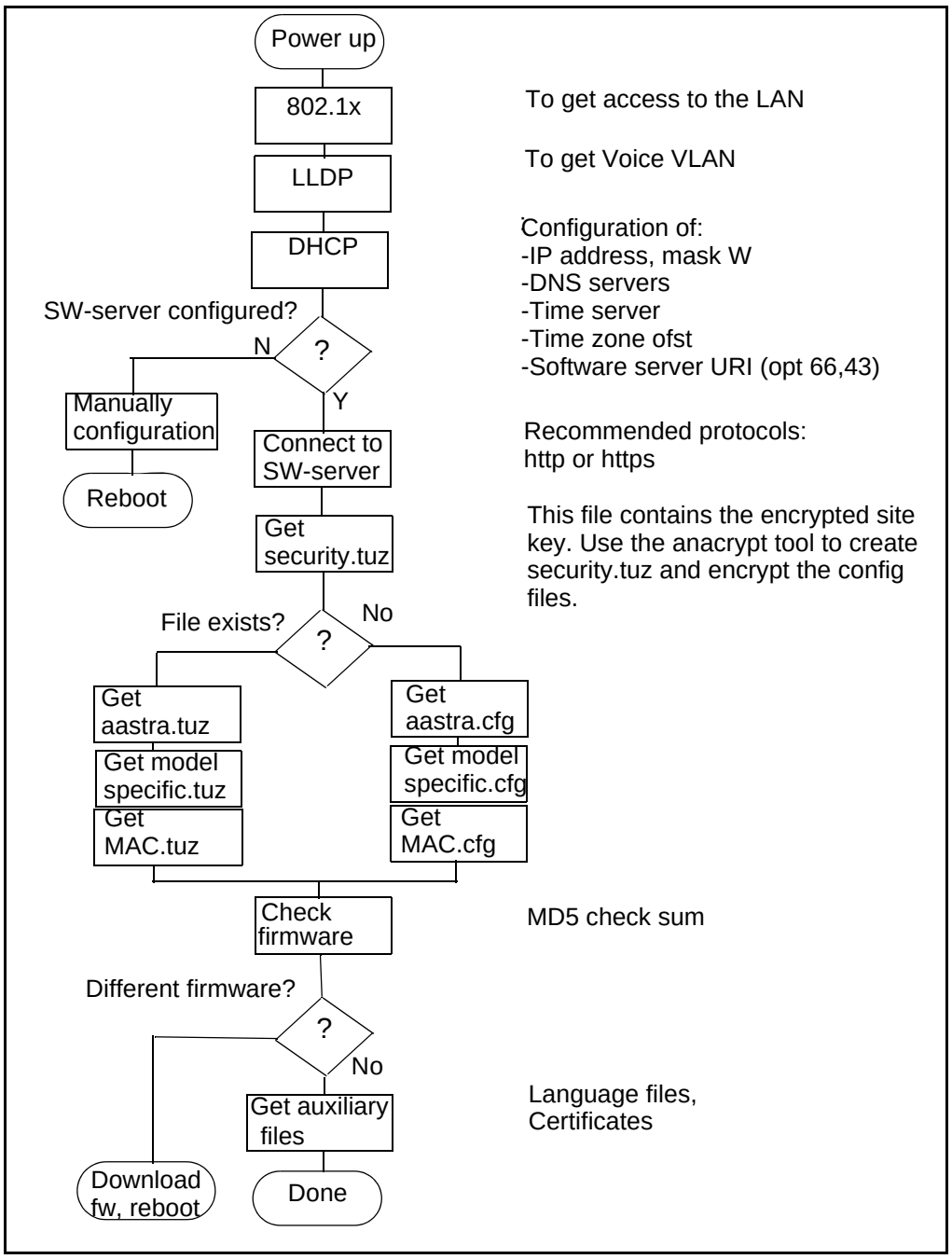
If Manager Telephony Server (MTS) is used and chapter 5.1 Install IP Phone SW Server on page 9 has been followed, the phone configuration files are generated and stored on the Software Web Server.

If MTS is not used, a software server must be set up supporting any of the protocols HTTP or HTTPS to host the phone firmware, language files and configuration files. The firmware files can be downloaded from Service Support Plaza. There are phone configuration template files adapted for MX-ONE stored under: **etc/opt/eri_sn/aastraSIPPhone**.

All firmware files are described in section 8.1 Phone software and configuration files on the software server on page 24.

When the phone is powered up, it will look for software (firmware) and configuration files on the software server according to its configuration server settings, see section11.8 Setting the IP Address and Download Protocol of the Software Server on page 34.

7.1 Boot flow chart



7.2 Connecting the Phone to a Network

To be able to connect the phone to a network, the following parameters must be configured:

- **The phone's IP address**, **subnet mask**, and **default gateway**. When using DHCP, these parameters are configured automatically.
- **The IP address of the software server**. This address is configured automatically using DHCP, or manually from the phone. If DHCP is used for providing this parameter, the DHCP server must be configured before the phones can connect to the network. For information on how to configure the DHCP server for providing the phone with the IP address to the software server, see 18.1 Data from DHCP on page 71.
- **The IP address of the SIP proxy / SIP registrar**. This address is configured using the configuration file or manually from the phone. For information on how to configure the phone with the IP address of the SIP proxy / registrar see 11.9 Setting the IP Address of the SIP proxy / registrar on page 35

7.3 Log on / Log off

To get log on / log off working, the configuration file shall be created in MTS and if this is not possible it shall be based on the **aastra.cfg** template which is stored in MX-ONE under **/etc/opt/eri_sn/aastraSIPphones.**

There are a number of options when register the telephone to MX-ONE:

- **Soft key / Programmable key**. The user can log on by pressing the log on key and enter the extension number and PIN code (if used). This is the recommended method.
- **Procedure: *11*PIN*extension number#**. The user can log on by entering this procedure. This method can be useful in a home office scenario.
- **MAC configuration file, <mac>.cfg**. This option can be useful for telephones where log on/off are not wanted, for example in reception area or in elevators.

  The extension number and the PIN code (if used) are defined in the MAC configuration file. The telephone will always be logged on. The user cannot log off the telephone. This can be useful for telephones in the reception area or in elevators. For settings in this file, see 8.1 Phone software and configuration files on the software server on page 24.

- **Web UI**. The extension number and the PIN code (if used) are defined via the web user interface. The telephone will always be logged on. The user cannot log off the telephone. To set the extension number and PIN via the web UI:

  Advanced Settings > Line 1 > Basic SIP Authentication Settings
  Fill in the Phone Number and Password (PIN code if this shall be used)

- **Telephone UI**. The extension number and the PIN code (if used) are defined in the telephone user interface. The telephone will always be logged on. The user cannot log off the telephone. To set the extension number and PIN from the phone UI:

  Press (Options key) **>Administrator Menu > SIP Settings > User Name** and **Password** (if PIN code is used)

7.3.1 Log on with Soft key

The procedure for logging on to MX-ONE is described in the Quick Reference Guide for each telephone model.

The necessary settings in the configuration files for this are created automatically when using MTS. In the configuration file **aastra.cfg** the following parameters must be enabled:

dynamic sip:1
*sip line1 user name: “Not configured” *)*
sip proxy ip: 192.168.10.10
sip proxy port: 5060
*sip registrar ip: 0.0.0.0 **)*
sip registrar port: 5060

*action uri startup: http://$$PROXYURL$$:22222/Startup?user=SIPUSERNAME$$ ***)*

*) this line is used in the case of emergency calls when the telephone is not registered. It is also used in the case of register with the *11 procedure.

**) shall be set to 0.0.0.0 because the system will replace the zeros with the actual IP address to the registrar as a part of the registration procedure.

***) the terminal will show the prompt with user and password at startup of the telephone.

In the **<model>.cfg** file, the following parameters must be enabled:

softkey5 label: “Log on”
softkey5 type: xml
softkey5 value: http://$$PROXYURL$$:22222/Logon

softkey5 states: idle
softkey5 line: 1

7.3.2 Log on with procedure

This method can be useful in a home office scenario when the registration towards MX-ONE is done via a firewall that only allows SIP signaling but not http signaling.

The procedure to register the telephone is:

- Enter ***11*PIN*extension number#**. The PIN code must be entered if the PIN code is initiated in TSE.
- If the registration is successful, the extension number and the name of the user is shown in the display.

The procedure to log off the telephone is:

- Enter **#11#**.
- If the log off is successful, the display shows **Logged off.**

7.3.3 Change of PIN code

The user can change the PIN code by entering the procedure *74*old PIN*new PIN#.

Note: If a <mac>.cfg file is used, the PIN code must manually be changed also in this file.

7.4 Log On/Off Key Not Used

If free seating shall not be used or if the log on/off key must be used to other features, the MAC configuration file can be used for deploying the phone number and PIN code in the telephones.

7.4.1 MAC configuration file

There are information in the template file for the mac file stored under **/etc/opt/eri_sn/aastraSIPphones**.

The MAC configuration shall be used for exceptions to the general key layut choosen for each model. See Chapter “Default key layout” if the general layout shall be changed.

Alternative1. Use MAC file to only logon via the logon prompt at bootup, which is set via “sip action uri startup” and not configure any logoff key.

The terminal can be logged of via command extension_unregistration --forced. Then it will prompt for logon after the reboot:

Write protect the logoff keyset.

Example: Change the logoff key to be a speedial to call the operator and write protect it by prefixing with “!”.

!softkey9 type:speeddial
!softkey9 label:Operator
!softkey9 value:09

The Diversion key kan also be write protected in the similar way.

As the ‘action uri startup’ is used Free Seating is enabled, which means that the terminal may be pushed out

Alternative2. Use the MAC file to explicitly set an extension number:

sip line1 user name:<extension number>
sip line1 password:<PIN code>

If the PIN code is changed in MX-ONE, it must also be manually changed in the <mac>.cfg file.

Inactivate action uri startup (set in aastra.cfg) by setting it to an empty string (this will also inactivate Free Seating. The terminal will not be pushed out when someone logs in using the same extension number on another phone):

action uri startup:””

With logon at startup disabled, SIP registrar must be set in the <mac> file to trigger the phone to register. (aastra.cfg setting is 0.0.0.0):

sip proxy ip: <mx-one ip address/host>
sip registrar ip: <mx-one ip address/host>

Change and write protect the logon/logoff key as described earlier.

7.5 Message Waiting Indicator

The red lamp in the upper right corner is called message waiting indicator and is used in the following cases:

- message waiting indication: blinking slow
- incoming call: blinking fast
- no service: lit.

8 Managing IP Phone SW

8.1 Phone software and configuration files on the software server

If any configuration file or firmware is changed on the software server, the phones are updated when restarted. The following files need to be stored on the software server:

<phone model>.st

The application firmware (software) for the phones. The names of the application files are:

- **6730i.st, 6731i.st, 6735i.st 6737i.st and 6739i.st**
- **53i.st, 55i.st and 57i.st (for 6753i, 6755i and 6757i)**

aastra.cfg

This file contains the configuration parameters for all 6700 phones in the system. The configuration file has to be adapted for each installation.
This file is created in IP Phone Configuration File task in Manager Telephone Server (MTS). If it is not possible to use MTS, the **aastra.cfg** template must be used which is stored in MX-ONE under **/etc/opt/eri_sn/aastraSIPphones/**.

aastra.tuz

This is the encrypted aastra.cfg file. The phone uses http protocol to fetch this file. The configuration file has to be adapted for each installation and then it has to be encrypted, see section 19.1 Encrypted Configuration Files on page 77.

<phone model>.cfg

This file contains configuration parameters for the key layout for each phone model. The names of the configuration files are:

- **6730i.cfg, 6731i.cfg, 6735i.cfg, 6737i.cfg, 6739i.cfg, 6753i.cfg, 6755i.cfg, 6757i.cfg**

The settings in <phone model>.cfg will override the settings in aastra.cfg.

This file is created in IP Phone Configuration File task in Manager Telephone Server (MTS). If it is not possible to use MTS, the **<phone model>.cfg** template must be used which is stored in MX-ONE under **/etc/opt/eri_sn/aastraSIPphones/**.

<phone model>.tuz

This is the encrypted model specific configuration file. The configuration file has to be adapted for each installation and then it has to be encrypted, see 19.1 Encrypted Configuration Files on page 77.

<mac>.cfg

When this file is used, it is possible to get unique parameter settings per telephone. This file is optional and the file looks similar to the **aastra.cfg** file. <mac> represents the mac address of the phone. Example: **00085D1B5D8.cfg**
The settings in <mac>.cfg will override the settings in aastra.cfg and in <phone model>.cfg.

The **<mac>.cfg** template must be used which is stored in MX-ONE under **/etc/opt/eri_sn/aastraSIPphones/**.

When deploying extension number and PIN code via this file, see section 7.4.1 MAC configuration file on page 22.

<mac>.tuz

This is the encrypted mac address configuration file. The configuration file has to be adapted for each installation and then it has to be encrypted, see 19.1 Encrypted Configuration Files on page 77.

lang_<nn>.txt

This file contains the display text in the specific language. <nn> can be de (German), es (Spanish), es_mx (Mexican Spanish), fr (French), fr_ca (French Canadian), it (Italian), pt (Portuguese), pt_br (Brazil Portuguese) and ru (Russian). It is also possible to create additional language files for other languages.

8.2 Installing the Firmware / Configuration files

When the phone starts, the phone fetches the configuration file from the software server and load new firmware if the application file on the software server differs compared to the one stored in the phone.

To force the phones to read the configuration files and to restart the phone if necessary, there are a number of options:

MX-ONE command

- extension_unregistration. If the **forced** parameter is used, the local configuration settings are cleared. The terminal has to be manually logged on with extension number and PIN code.

MTS

- **Telephony > IP Phone > Unregistration**

Phone UI

- **Options** > **Restart Phone**

Web UI

- Log in to the web interface. Click on **Operation** > **Reset** > **Restart Phone**

8.3 Firmware upgrade

Firmware upgrade can be done in one of the following ways:

- Web UI: **Advanced settings** > **Firmware Update**
- The phone will automatically look for firmware update and configuration files during the boot process.
- Define in the configuration file **aastra.cfg** if and when phones shall check for new firmware and changed configuration file. Both registered and not registered terminals will be updated. Example from the configuration file:

  auto resync mode: 3 #Check for new fw and configuration files every day.
  auto resync time: 03:00 #The scheduled time
  auto resync max delay: 60 #Specifies the maximum time, in minutes, the phone waits past the scheduled time before starting a resync.

- MX-ONE command: **extension_unregistration**.
- MTS: **Telephony > IP Phone > Unregistration**

8.4 Viewing Software Version

It is possible to display the versions of the software units.

Phone UI

- Select **Options** > **Phone Status** > **Firmware info**.

Web UI

- Select **Status** > **System Information** > **Firmware information**.

MX-ONE

- MX-ONE command: **extension_info**

9 Restart / Restore

There are three options:

- Restart the phone. Can be used when settings shall be applied.
- Remove local configuration settings. The settings that are done from the phone UI and web UI are lost.
- Restore to factory default. The phone gets the same data as when leaving the factory and removes any saved directory files.

9.1 Restart

Phone UI

- Press the **Options** key
- Scroll down and select **Restart Phone**

Web UI

- Log in to the web interface. **Operation** > **Reset** > **Restart Phone**

9.2 Remove local configuration settings

All configuration made on the phone, via Web UI or configured by MX-ONE at logon is stored as local configuration in the phone. By removing the local configuration the administrator can ensure that phone is configured according to configuration files only.

Phone UI

- Press the **Options** > **Administrators Menu** > **Erase Local Config** Restart the phone.

  6739i; Press **Options** > **Advanced** (log in as administrator) > **Reset** > **Erase Local Config.**

Web UI

- Log in to the web interface. Click on **Operation** > **Reset** > **Remove Local Configuration Settings.**
  Restart the phone.

It is also possible to remove the local configuration settings registered terminals, by entering the following command from MX-ONE:

extension_unregistration with parameter **forced**

9.3 Restore to Factory Default

Factory default reset will force the phone to go back to the initial setting. If configuration server is not set via dhcp options, you will need to set it again. If the configuration server shall be accessed via https, only the commercial root CAs (Verisign etc) are preloaded.

Phone UI

- **Options** > **Administrators Menu** > **Factory Default**

  6739i; Press **Options** > **Advanced** (log in as administrator) > **Reset** > **Factory Default.**

Web UI

- Log in to the web interface. Click on **Operation** > **Reset** > **Restore to Factory Defaults**

10 Entering Administrator Mode

Phone UI

- Press (options) key
- Scroll down and select **Admin Menu**

  6739i; select **Advanced**.
- Enter the administrator password: 22222 (which is the default password but can be changed)

Web UI

- Find the IP address of the telephone by selecting **Options** > **Phone Status** > **IP & MAC Addresses**
- Enter the IP address to the telephone into the address field in the web browser in the PC and press enter.
- Log in to the web interface by enter
  User name: **admin**
  Password: **22222** (which is the default password)

11 Configuring the Phone

This chapter describes how to configure the phone from the phone menus as an administrator.

This chapter also covers the configuration via the configuration files, **aastra.cfg**, **<model>.cfg** or **<mac>.cfg**.

The parameters can be set in any of these configuration files, but in this section it is the recommended placing that is described. If one parameter occurs in several configuration files, it is always the last read parameter value that the telephone uses.

11.1 Settings mode

To enter into settings mode in the phone user interface:

- Press (Options key)

To enter the web user interface:

- Find the IP address of the telephone by selecting **Options** > **Phone Status** > **IP & MAC Addresses**
- Enter the IP address to the telephone into the address field in the web browser in the PC and press enter.
- To log in to the end user page, see 12.2 Web Interface Passwords for End Users on page 63
- To log in to administrator page, see 10 Entering Administrator Mode on page 29.

11.2 Settings in the configuration file aastra.cfg

The necessary settings in the configuration file for getting the telephones to work in a correct way with MX-ONE are created automatically when using MTS. In the configuration file **aastra.cfg** the parameters must have the following values:

! sip aastra id: 1
! sip send line: 1
! sip xml notify event: 1
! sip pai: 1
! directed call pickup: 1

! directed call pickup prefix: Pickup
! collapsed context user softkey screen:1

softkey selection list: "none,speeddial,line,xml,speeddialxfer,speeddialconf,phone-lock,empty" *)

*) the options for the softkeys that are working with MX-ONE and is presented in the web UI.

11.3 Automatic LAN Access Control, IEEE802.1x

The IEEE802.1x standard is used for port access control authentication. The LAN switch must support IEEE802.1x signalling and there must be a RADIUS server handling the authentication. This feature supports both EAP-MD5 and EAP-TLS protocols.

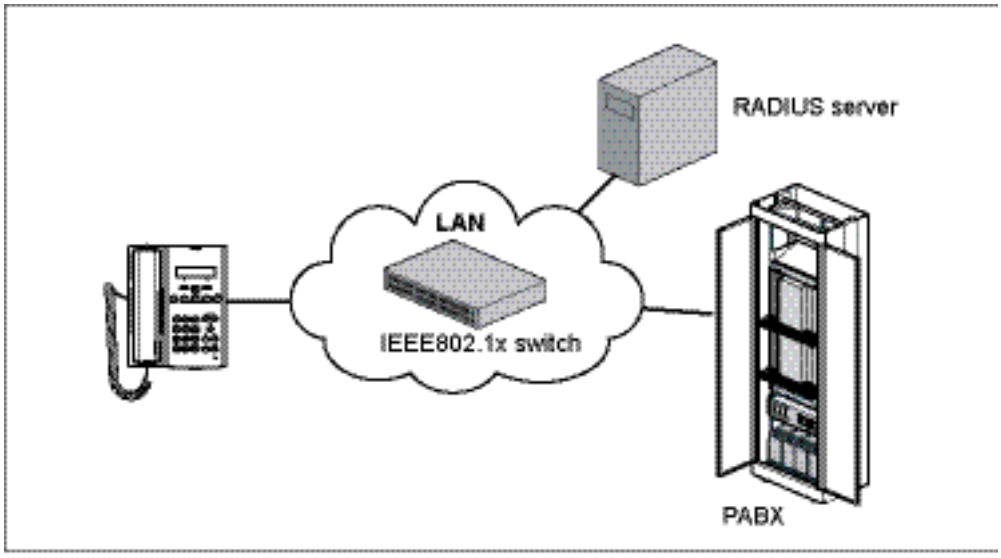


Figure 3: Components in LAN access control

Below is an example of the settings in the phone configuration file (aastra.cfg) when EAP-TLS shall be used:

eap type: 2
identity: Phone_Floor1
802.1x root and intermediate certificates:aastra67xxi/Aastra_Client_ca.pem
802.1x local certificate:aastra67xxi/Aastra_Client_cert.pem
802.1x private key: aastra67xxi/Aastra_Client_key.pem
802.1x trusted certificates: aastra67xxi/Aastra_Client_ca.pem

The certificate shall be available on the software server. In the example above they are stored under the folder *aastra67xxi*. The certificate files must be loaded into the phones before IEEE802.1x is activated.

Below is another example showing how to set the parameters in aastra.cfg when MD5 shall be used:

eap type: 1
identity: Phone1
md5 password: Anypass

LAN switch

Below is an example how to configure a Cisco switch to enable IEEE802.1x:

aaa authentication dot1x default group radius

dot1x system-auth-control
radius-server host X.X.X.X auth-port 1812 acct-port 1813
radius-server key XXX

Configuration of an access port for IP telephony:

interface FastEthernetx/0/x
description Aastra accessport
switchport mode access
switchport voice vlan 150
authentication host-mode multi-domain
authentication port-control auto
authentication periodic
authentication timer reauthenticate 120
authentication violation protect
dot1x pae authenticator
auto qos voip trust
mls qos trust dscp
spanning-tree portfast

Radius server

In the Radius server the certificate matching the one in the phones must be available (when EAP-TLS is used). It is also important to configure the port to enable the telephony VLAN otherwise the telephone will try to use the data LAN.

In the example below the settings for enabling of telephony VLAN in the configuration file for a Radius server from FreeRadius is shown when using a Cisco LAN switch:

Phone_Floor1 Cleartext-Password := "GJM"
cisco-avpair == "device-traffic-class=voice"

In combination with the examples above (parameters marked with red) this will mean the telephones with the identity Phone_Floor1 will use VLAN 150.

For more information about how to set up IEEE802.1x in the phones, see Administrator Guide for Aastra Models 6700i and 9000i Series IP SIP Phones.

11.4 LLDP-MED

The telephones have support for Link Layer Discovery Protocol for Media Endpoint Devices (LLDP-MED), which can for example be used to get the VLAN identity or the emergency location identification number (ELIN). In this section it is only the VLAN identity that is described. For information about ELIN, see Administrator Guide for Aastra Models 6700i and 9000i Series IP SIP Phones.

Note: If LLDP is not used in the network, LLDP can be disabled in the aastra.cfg file, which means that the telephone will start much faster.

Previously, Aastra IP Phones had a 5 second timer for listening to LLDP-MED responses when the phone is booting up. If LLDP-MED responses are received after this initial listening period, the phone will not get access to the telephony VLAN. If there is an untagged LAN, the phone will use it and may be hanging in a DHCP negotiation.

Dependant on when the phone was manufactured, this problem can still occur at new installation. If this problem occurs, the recommendation is to set the timer in the LAN switch temporarily to 5 seconds, start the phones with an aastra.cfg file where the time (parameter *lldp startinterval*) is changed to match the time in the LAN switch at ordinary operation for example 32 seconds. When the phones are started, the timer shall be changed back to the original value in the LAN switch.

The example below shows which parameters to set in aastra.cfg:

 # LLDP enabled = default
lldp: 1
 # LLDP update interval 30 s
lldp interval: 30
 # Controls the LLDP start interval, 32 s
lldp startinterval: 32

The parameter *lldp startinterval* is only valid during the phone bootup process and it will control the LLDP time-out interval where the phone sends LLDP advertisements and listens for the LLDP responses from the switch before proceeding to the DHCP stage. The default value of this parameter is 32 seconds.

For more detailed information see Administrator Guide for Aastra Models 6700i and 9000i Series IP SIP Phones.

11.5 Enabling / Disabling DHCP

Follow the steps below to enable or disable DHCP:

Options > **Admin Menu[**6739i; select **Advanced]** > **Network Settings** > **DHCP Settings** > **DHCP**

11.6 Setting the Phone's IP Address

If DHCP is used, the phone's IP address is set automatically, using the DHCP server. To be able to set the phone's IP address manually, DHCP must first be disabled on the phone, see 11.5 Enabling / Disabling DHCP on page 33.

Options > **Admin Menu [**6739i; select **Advanced]**> **Network Settings** > **IP Address**

11.7 Setting the IP Address to the Default Gateway

If DHCP is used, the IP address to the default gateway is set automatically, using the DHCP server. To be able to set the IP address to the default gateway manually, DHCP must first be disabled on the phone, see11.5 Enabling / Disabling DHCP on page 33.

Options > **Admin Menu[**6739i; select **Advanced]** > **Network Settings** > **Gateway**

11.8 Setting the IP Address and Download Protocol of the Software Server

To download the phone software and configuration files, the phone must be configured with the type of protocol and IP address matching the software server (configuration server). The configuration server can be set using these alternatives:

- Manually from the phone UI; **Options** > **Admin Menu** > [6739i; select **Advanced]**> **Configuration Server**.
- Manually via the administrator web UI: Click on **Advanced Settings** > **Configuration Server**
- Automatically using DHCP, see 18.1 Data from DHCP on page 71.

11.9 Setting the IP Address of the SIP proxy / registrar

The phone is configured with the IP address of the SIP proxy using one of the following methods:

1. In the configuration file **aastra.cfg** in the parameter: **sip proxy ip**. The necessary settings in the configuration file for this are created automatically when using MTS.
2. In the configuration file **<mac>.cfg** in the parameter: **sip proxy ip**.
3. Phone UI: **Options** > **Admin Menu** > **[**6739i; select **Advanced]** > **SIP Settings** > **Proxy IP/Port**
4. Web UI: Click on **Advanced Settings** > **Global SIP** > **Basic SIP** > **Basic SIP Network Settings**

11.10 Using Virtual LAN (VLAN)

The following VLAN data can be set:

- Enable VLAN tagging
- VLAN identity

The following configuration alternatives are available:

- aastra.cfg file.
- Link Layer Discovery Protocol for Media Endpoint Devices (LLDP-MED).
- In DHCP option 43, see section 18.3 DHCP Settings for Option 43 and 60 on page 72.
- Phone UI. **Options** > **Admin Menu** > **[**6739i; select **Advanced]** > **Network Settings** > **Ethernet & VLAN - VLAN Settings**
- Web UI: Click on **Advanced Settings** > **Network** > **VLAN**

For detailed configuration information, see Administrator Guide for Aastra Models 6700i and 9000i Series IP SIP Phones

11.11 Setting Time and Date

Time and date are set via the NTP protocol from a time server.

The time and data format is also possible to change.

The following configuration alternatives are available:

- aastra.cfg file.
- Phone UI. **Options** > **Preferences** > **Time and Date**
- Web UI: Click on **Preferences** > **Time and Date Settings**

It is possible to use LIM 1 in MX-ONE as a NTP server.

11.12 Configuring Language Settings

The language for the display texts and the language for the input via the key pad can be set.

English is always available in the telephone and cannot be removed. It is possible to add more languages via the configuration file and to define the default language. Example:

lang 1: lang_de.txt
lang 2: lang_fr.txt
lang 3: lang_es.txt
lang 4: lang_sv.txt
language: 4

In the example, English, German, French, Spanish and Swedish will be shown in the phone menu with Screen Language and the default language will be Swedish.

See also, section 8.1 Phone software and configuration files on the software server on page 24 and section 7.3.1 Log on with Soft key on page 21.

Change the language in a telephone by:

- Phone UI. **Options** > **Preferences** > **Language**
- Web UI: Click on **Basic Settings** > **Preferences** > **Language Settings**

Some text strings are sent out to from the PBX to the telephone. To order the PBX to send out the right language enter from the telephone:

***08*n#** where n is the language number in MX-ONE.

The labels for *Message Waiting* and for *CorpDir* must be translated by the system administrator, by using Manager TS to change the label for this key, or by editing the model specific configuration file for each phone model.

11.13 Using Shortcut Keys

Shortcut keys can be of two types:

- **System keys**. Keys that are common on all terminals within a certain model. Example: log on/off, diversion, message waiting, corporate directory, etc.
- **Individual keys**. Keys that are unique for each user. Example: speed dial, monitoring keys, extra directory number etc.

11.13.1 Key numbering

The numbering of the keys for the different models are shown in the Administrator Guide for Aastra Models 6700i and 9000i Series IP SIP Phones.

There are the following types of keys that can be used as shortcut keys:

- Softkeys. In 6737i/57i, 6735i/55i and in 6739i.
- Top Softkeys. In 6737i/57i.
- Programmable keys (hard keys). In 6730i/31i, Aastra6753i, 6735i/55i.
- Expansion module key, see section 14 Expansion Modules on page 65.

The **softkeys** are reserved in the following way:

- **1-4**, system keys. For MX-ONE busy services (call back call pickup etc.) and for xml kit applications. These are predefined as key type XML in aastra.cfg and will get the actual key label and value when busy services is offered.
- **5**, system key. Logon/Logoff is the first key visible in idle state.
- **6**, system key. For Corporate Directory search (if this feature shall be used).
- **7,** system key, For Diversion in 6739i. In the other terminal models can softkey 7 be used for system key functions which shall be equal on all telephones in a model, see 11.13.2 Default key layout on page 39.
- **8**, system key to be defined by the system administrator, see 11.13.2 Default key layout on page 39.
- **9 and upwards**, individual keys. The following features are available and programmed form the PBX:
  - BLF keys (MNS keys)

– Speed dial (TNS) key. Can also be programmed from a menu in the telephone or from the web interface in the phone.

– Personal Number (PEN) key

– Shared Call Appearance (SCA) key

– Extra Directory Number (EDN) key

– Malicious Call Trace (MCT) key

The connection between the soft key number in the phone and the logical key number when initiating a key in MX-ONE is: The logical key number 1 corresponds to soft key number 9 in the phone, logical key 2 to softkey 10, etc.

The first available key number that can be used as an individual key is defined in the configuration file in MX-ONE: */etc/opt/eri_sn/ip_telephony.conf*

Softkeys of the type *none* are not shown in the display. For example; if the softkey with key number 10 is programmed with a speed dial number, the softkey will show up in the telephone display on the first key position with type equal to *none*. If the user want to have the key on the phone display on the same position as where the key was programmed, the softkeys must be set to type equal to *empty*.

The **top softkeys** are reserved in the following way (only 6737i/57i):

- **1-5**, system keys, see 11.13.2 Default key layout on page 39.
- **6-10**, individual keys of the type: speeddial, xml, none, empty speeddialxfer, speeddialconf or phone lock.

Table 3 Number of Shortcut keys

Model	Hard keys	Soft keys	Comment
6730i/ 31:	8 top	-	
6739i:	-	55	
6753i:	6		This model must have an expansion module if MNS, TNS (initiated from the PBX) MCT, PEN shall be used.
6735i/ 55i:	6 top	6 bottom	20 bottom softkeys on 4 pages.
6737i/ 57i:	-	6 top, 6 bottom	20 bottom softkeys on 4 pages. 10 top softkeys on 2 pages.

Expansion modules can be added to 6753i, 6735i/55i, 6737i/57i and 6739i.

When adding an expansion module to the telephone, the keys for MNS, TNS (defined in the PBX), MCT and PEN are moved from the telephone to the expansion module.

11.13.2 Default key layout

This chapter shows the default key layout per model. The following system keys can be changed or removed via Manager TS or via editing the model specific configuration files.

- Services
- Local Directory
- Callers List
- Message Waiting
- Corporate Directory

If one of these functions is removed, it can be replaced by a key with another function that shall be generic for all phones of a certain model. The following options are possible to set:

- **speeddial**
- **xml**
- **speeddialxfer** (the softkey is configured to transfer calls and configured for speed dialing to a specific number.)
- **speeddialconf** (the softkey is configured as a speed dial key and as a conference key.)
- **phone lock** (the key is used to lock / unlock the phone).
- **none** (the softkey is not used)
- **empty** (the softkey is configured to force a blank entry on the phone display

The following keys that are pushed out from MX-ONE Telephony Server and can not be changed via Manager TS, they need to be edited manually in config file **/etc/opt/eri_sn/ip_telephony.conf.**. (If the key values are set to “”, they are not pushed to the phone). This file must be changed in all servers in the system. At upgrading of the MX-ONE software this file have to be edited again. ‘restart -u SIPLP’ is required after the file is edited to activate changes.

- Logon/Logoff
- Diversion

The first individual key index and key base (prgkey,softkey or topsoftkey) is also set via ip_telephony.conf

The <MAC>.cfg can be used for a phone to override the default key layout. You may decide that the a specific phone shall have an extension number which shall not be logged off and not to be used for free seating. See Chapter “Logon/Off Key Not Used”.

See also 8.2 Installing the Firmware / Configuration files on page 25.

The default key layout for the different models is shown below.

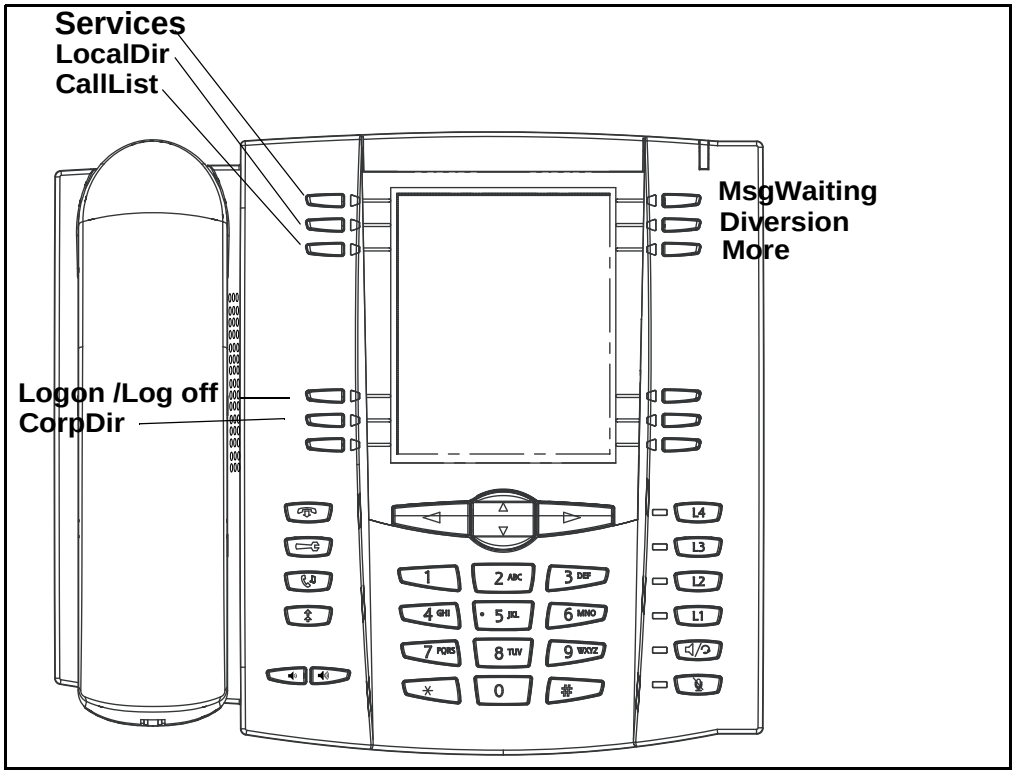


Figure 4: Aastra 6737i/57i

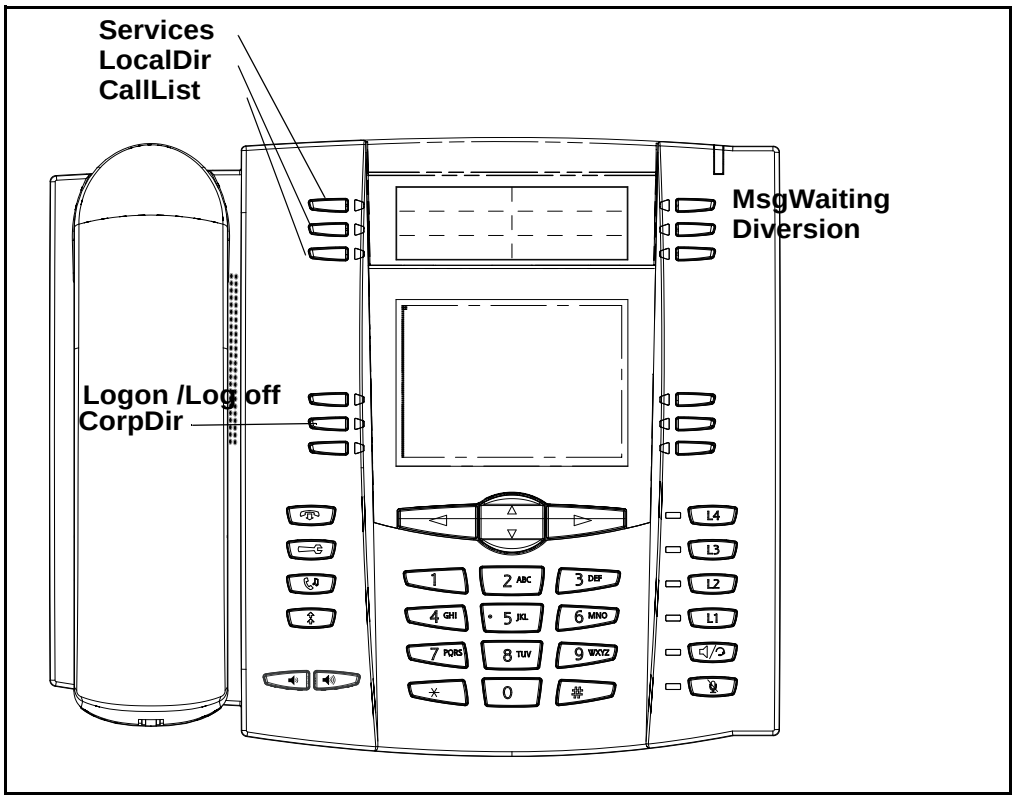


Figure 5: Aastra 6735i/55i

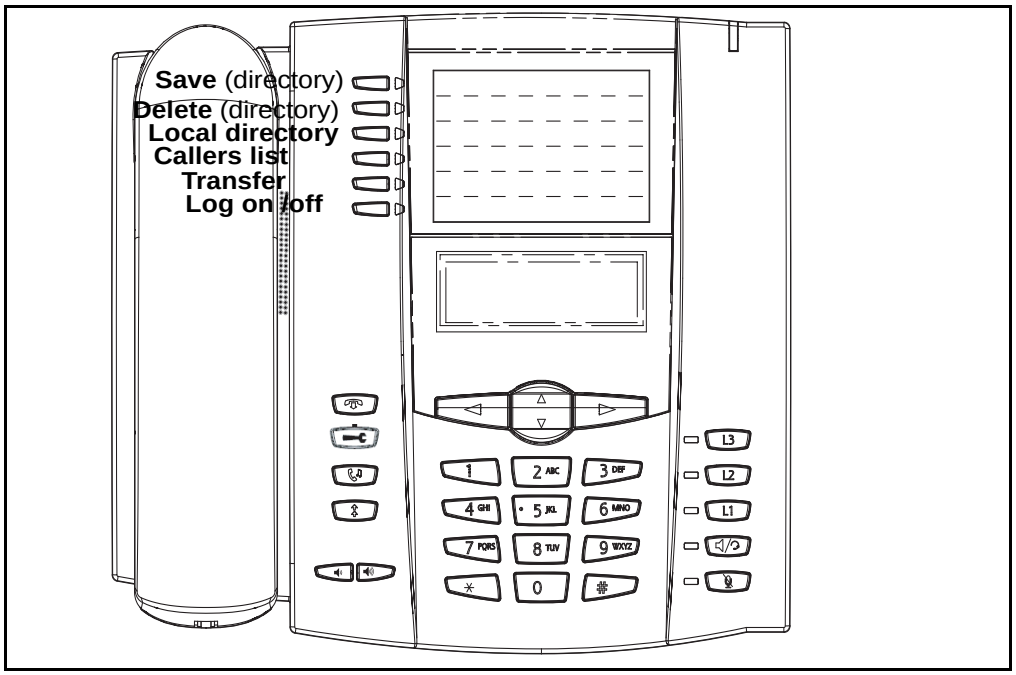


Figure 6: Aastra 6753i

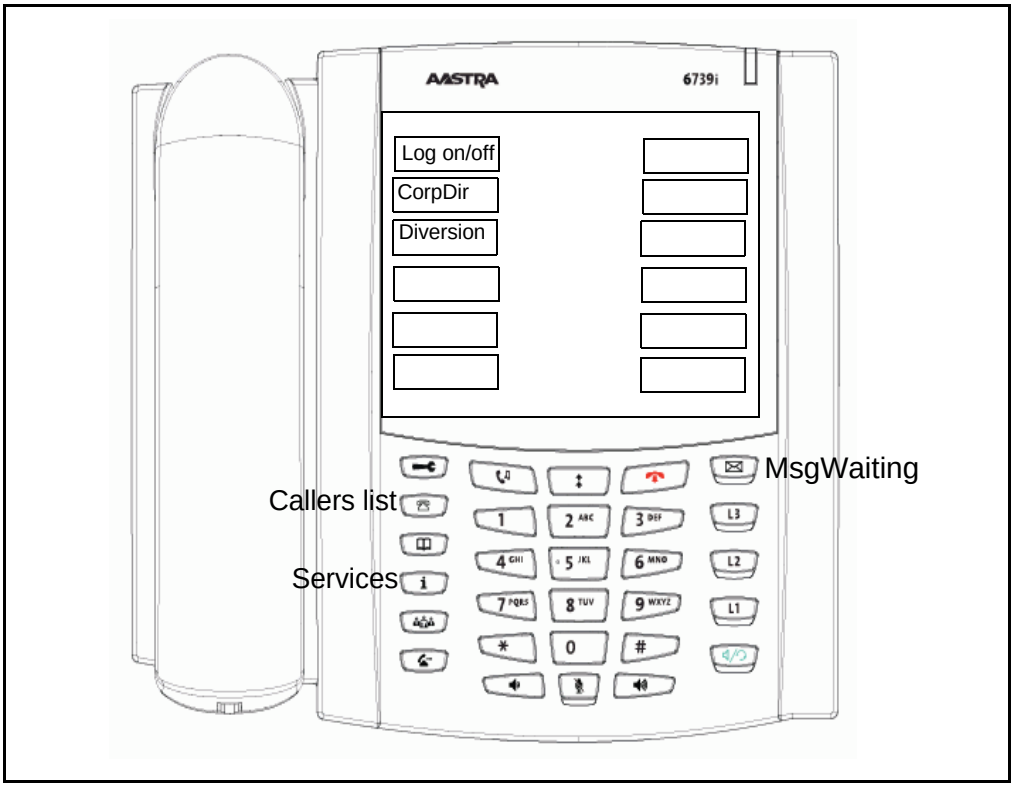


Figure 7: Aastra6739i

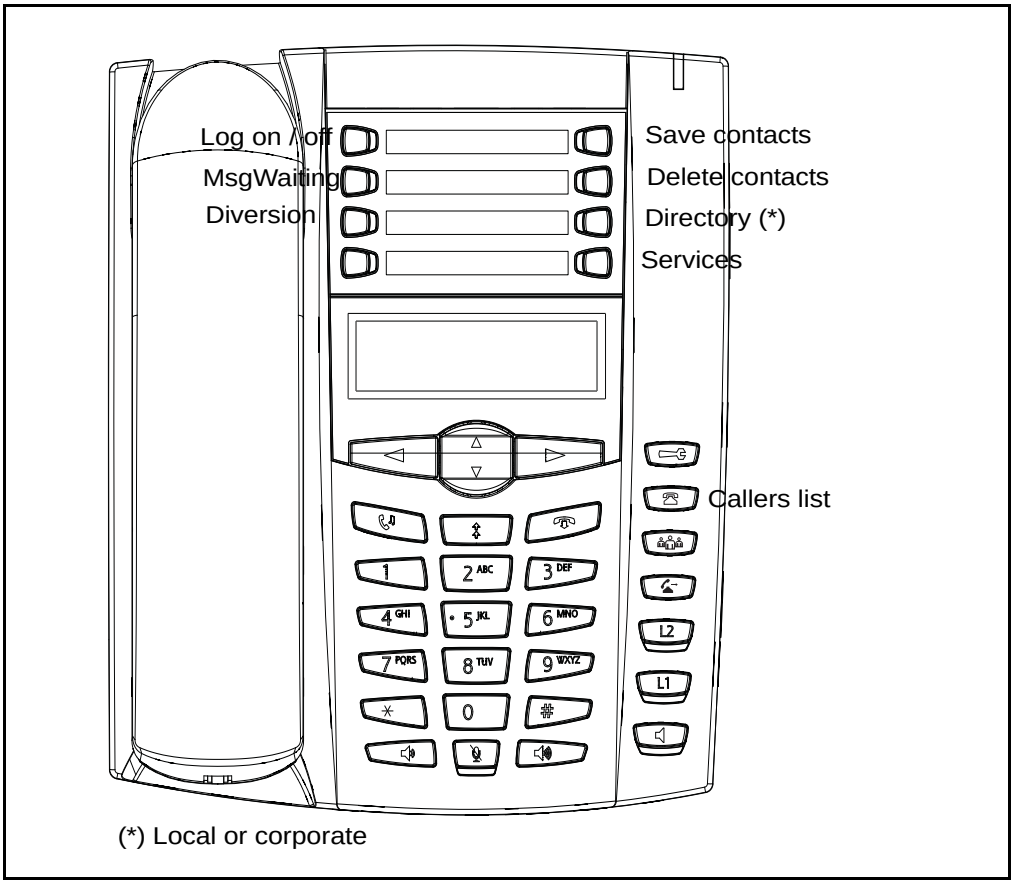


Figure 8: Aastra 6730/31

11.13.3 Flexible IP Function keys in MP

For most telephone models, Manager Provisioning makes keys available for individual programming based on the assumption that both ip_telephony.conf and the model specific configuration file has standard definitions. For the 6730i and 6731 models, Manager Provisioning analyzes the actual settings to determine which function keys are available for individual programming.

The function keys defined for an Aastra67xxi terminal depends on two configuration files:

- the global ip_telephony.conf
- model specific, e.g. 6731i.cfg.

The ip_telephony.conf file is stored in the MX-ONE Telephony Server and is read by it at start-up but not by the telephones directly. This file contains telephone model specific definitions of function keys that shall be pushed to the telephone, e.g. Log on/off and Diversion keys. Also a key offset is defined per model type which defines an offset number of the first key that can be programmed in the TS.

The model specific configuration file, e.g. 6731i.cfg, is read by the telephones and contains this model's default definition of the function keys layout. These files can be defined and changed in the Configuration File task in MTS, see chapter 6.

Example - Make all function keys programmable

In this use case all keys will be freed up for the user to be programmed individually.

Do as follows:

1. Edit **/etc/opt/eri_sn/ip_telephony.conf** in MX-ONE Telephony Server ,so it looks like the following example :

```
#6731i:KeyType={prgkey}
AastraTerminal:6731i:Model: "0E"
AastraTerminal:6731i:ExtensionKeyType: "prgkey"
AastraTerminal:6731i:ExtensionKeyOffset: 0
AastraTerminalKeys:6731i:LogonKey:pos: ""
AastraTerminalKeys:6731i:LogonKey:type: ""
AastraTerminalKeys:6731i:LogonKey:value: ""
AastraTerminalKeys:6731i:LogoffKey:pos: ""
AastraTerminalKeys:6731i:LogoffKey:type: ""
AastraTerminalKeys:6731i:LogoffKey:value: ""
AastraTerminalKeys:6731i:DiversionKey:pos: ""
AastraTerminalKeys:6731i:DiversionKey:type: ""
AastraTerminalKeys:6731i:DiversionKey:value: ""
```

Figure 9: Configure file

Note: Each Telephony Server holds a copy of the ip_telephony.conf file. For consistency and simplicity, any change to it should be made consistently on all telephony servers. Settings apply to all telephones of the same model: In the example above, the logon/logoff key is disabled for all 6731i telephones, and all keys are open for programming since the offset value is zero.

2. **Restart unit SIPLP,** to activate changes. Use command **restart -u SIPLP --lim x.**

3. Logon to the web interface of the phone. Unmarke (Disable) the option **DHCP download Options**, and save the options.

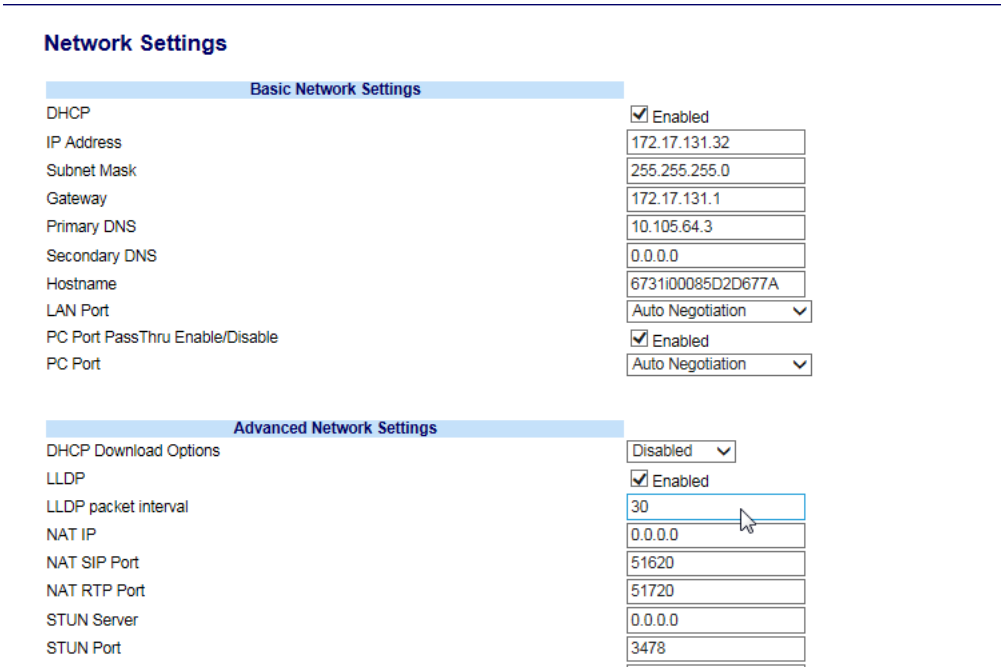


Figure 10: Network Settings

4. **Select** the right SW Server.

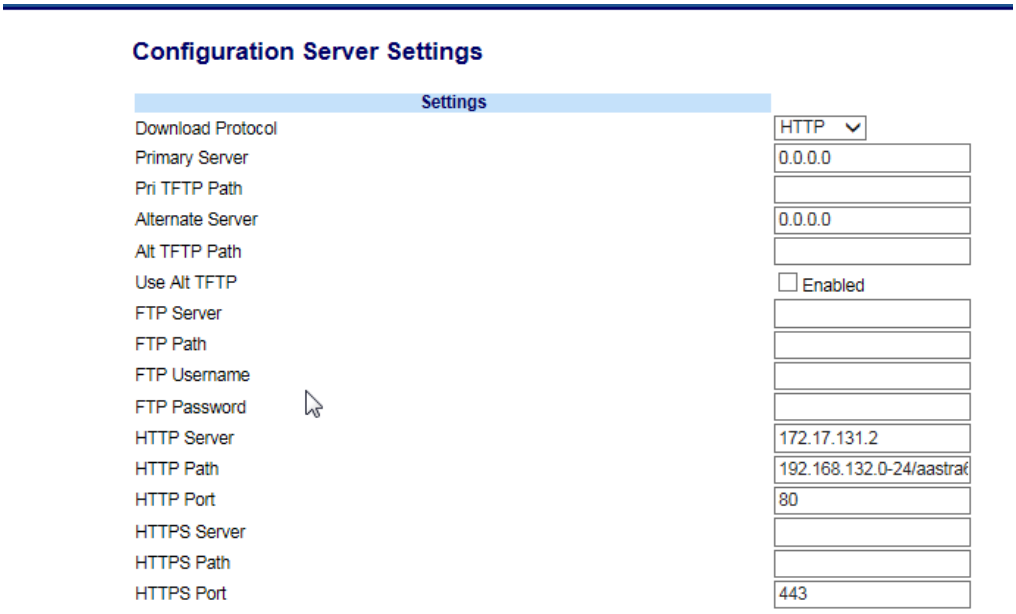


Figure 11:Configuration Server Settings

5. **Restart** the phone.
6. Open **MP**, and click **Telephony** tab. **Add a SW server** for your configuration files.
7. **Restart** a unit SIPLP, to activate changes, use command **restart -u SIPLP-lim x**.

Figure 12: SW Server

8. **Specify a IP Phone server and a domain folder**, if used.
   In MP the default IP Phone Server and Domain Folder has to be setup to point to the correct configuration file (the same as used by

the phone). This is done in the Subsystem task by selecting the IP Phone Server in the dropdown list and selecting one of the available folders in the Domain Folder dropdown list.The admin may also manually type in any IP Server address, at which the MP should read the model specific configuration files. This possibility is there in case there is no association between an MTS and an IPP Server that Manager Provisioning can retrieve or for environments where the IPP Server is not used to hold model specific configuration files.

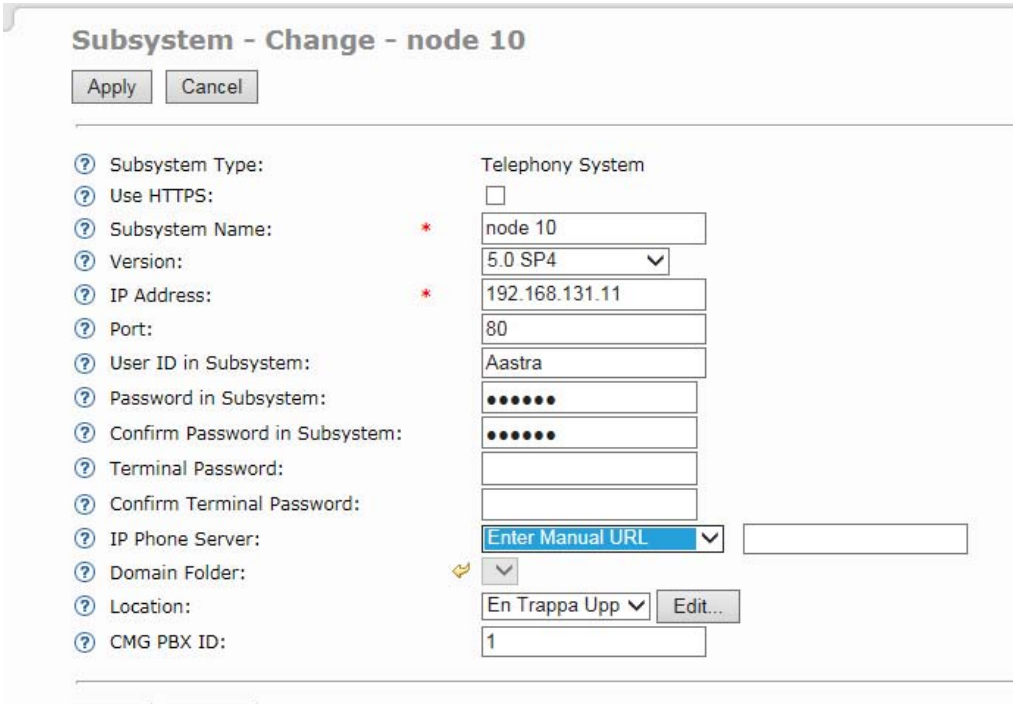


Figure 13:Specify IP Phone server and domain folder

9. **Start MTS, and edit** your configuration file.

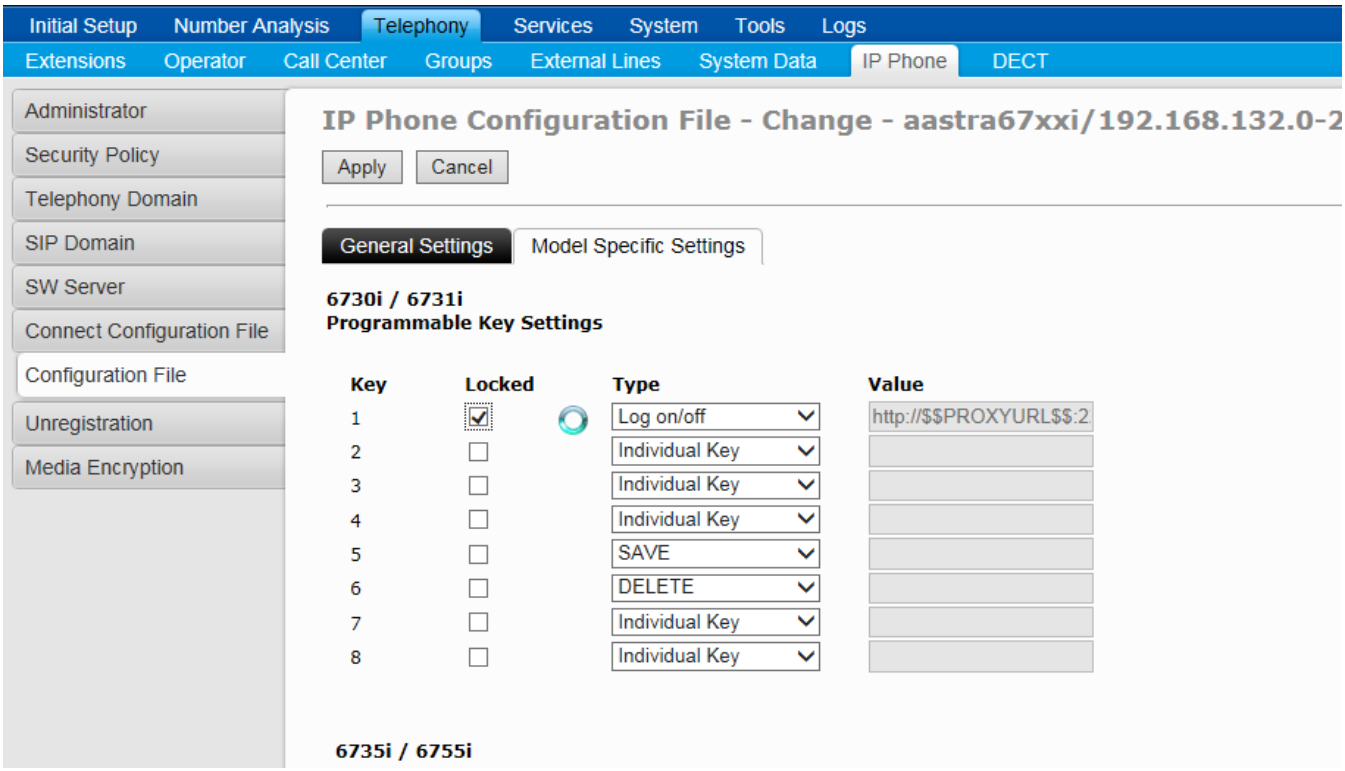


Figure 14:MTS settings, in this example keys 2-8 are flexible as key 1 is locked.

Note: If you want a different function keys layout on some special extensions even if they are using the same telephone model as the default extension, you can do this by:

1. Define a different model configuration file, e.g. 6731i.cfg, in Configuration File task in MTS, see chapter 6, and store it in another folder (Domain or Subnet) or on a different IP Phone Server.

2. Edit the special extensions in the Extension task, click the “Advanced” button and select the IP Phone Server and/or Domain Folder previously defined in the dropdown lists.

11.13.4 Assigning Phone Numbers to Speed dial keys

Shortcut keys that are not assigned to functions or monitored extensions can be assigned to phone numbers as speed dial keys (also called TNS keys). The user can initiate phone numbers to shortcut keys from the web UI.

When programming softkeys as speed dial keys from the web UI do not use key number 1-8 for bottom softkeys and 1-4 for top softkeys.

When a user has programmed a speed dial key the number and the label are stored locally in the phone and do not follow the user when he/she

logs on with another telephone. Speed dial data initiated via MX-ONE follows the user.

To program speed dial to a programmable key (hard key) that is empty from the phone UI:

1. Press the key for a couple of seconds until the input field appears in the display.
2. Enter the name and the number (or procedure with * and #).

Note: In 6735i/55i and 6737/57i the recommendation is to not use the phone UI for the programming of ***softkeys*** as speed dial keys because it will interfere with the keys used by the system. In 6739i it is ok when using the **Options key** > **Softkeys** to program the softkey with higher key number than 8.

To program a speed dial key (hard keys and soft keys) from the web UI:

1. Click on: **Operation** > **Softkeys and XML** or
   Operation > **Programmable keys** or
   Operation > **Expansion Module**
2. Enter the name and the number (or procedure with * and #).

Note: Do not program softkey 1-8 because these can interfere with keys used by the system.

To edit an existing speed dial key from the phone UI:

Options > **Preferences** > **Speed Dial Edi**t.

11.13.5 Assigning MNS Keys

The shortcut keys can be configured as monitoring keys (that is, assigned to monitor other extensions). The LED of the shortcut key is used for indicating the status of the monitored extension. By pressing the shortcut keys, calls to monitored extensions can be answered. Secretary functions is one example of this.

Monitoring keys are configured from the PBX.

To get the MNS key working, some parameters in the configuration file have to be enabled, see section 11.2 Settings in the configuration file aastra.cfg on page 30.

It is possible to change the type of ring signal (periodic, muted, visual only etc.) for the MNS key and it is changed from the PBX or from Manager Provisioning. The default value is visual only. No settings for this can be done in the aastra.cfg file.

It is also possible to set a pop-up option for monitoring keys. With this feature enabled, the page containing a monitor key is displayed when a

call to the associated monitored extension is received. This feature is enabled in the aasta.cfg file:

blf activity page switch:3

There are the following options:

- **0:** Page switching disabled. Default value.
- **1:** Switch page when the monitored extension receives a call.
- **2:** Switch page when the monitored extension receives a call or put the call on hold.
- **3**. Switch page when the monitored extension receives a call or put the call on hold or when the monitored extension change to speech state.

To avoid too much flashing of switching pages, all the MNS key should be collected on one page.

11.13.6 Shared Call Appearance (SCA)

The SCA feature allows a group of terminals to control the incoming and outgoing calls on a common line. The SCA feature is useful in work groups where it must be easy to exchange and move calls between the members.

The following terminal models can be initiated to have SCA lines: Aastra 6730i/31i, 6739i, 6735i/55i, 6737i/57i.

The extension number for a SCA line can be represented on a number of terminals. The main SCA extension number for a certain terminal is placed on L1 and L2. If the terminal shall monitor another SCA line, the hardkeys L3 and L4 are used. If additional monitoring SCA lines are initiated, they are represented on softkeys.

The reason to initiate two lines (e.g. L1 and L2) for each SCA number, is that if L1 is busy it is still possible to take another call on L2.

For a detailed description of the SCA feature in an MX-ONE environment, see feature description Shared Call Appearance.

For a description of the lamp indications when the SCA feature is used, see the Quick Reference Guide for each telephone model.

The SCA feature is initiated with Manager Provisioning (or via the MX-ONE command interface). No settings in the aastra.cfg file are needed.

11.13.7 Extra Directory Number (EDN)

One or several Extra Directory Numbers can be added to an existing extension which has an own directory number on Line1. The extra direc-

tory numbers are represented on line keys or softkeys. The EDN number has basically the same characteristics as the Line1 except for busy; when there is a call on an EDN line, the line is regarded as busy. When there is a call on Line1 it is still possible to receive another call on Line2.

The EDN keys are initiated with Manager Provisioning (or via the MX-ONE command interface). No settings in the aastra.cfg file are needed.

For more information how to initiate the softkeys as EDN keys, see 11.13 Using Shortcut Keys on page 37.

11.13.8 Softkeys for busy services

To get the softkeys for busy services to show up on the first page, one parameter in the aastra.cfg file has to be enabled:

! collapsed context user softkey screen:1

The following is valid for 6735i/55i and 6737i/57i: The busy services call-back and call pickup are shown on softkeys on page 1 when calling a busy extension. Call waiting and intrusion are shown on softkeys on page 2, which means that the user has to press **More** to see these soft keys.

The following is valid for 6739i: All busy services are shown on the first page.

See also section 11.2 Settings in the configuration file aastra.cfg on page 30.

11.13.9 Key Lock / Unlock

It is possible to lock or unlock softkeys, programmable keys and expansion keys. When key locking is enabled, the phone uses the settings from the configuration files and ignores any previous local configuration. A user cannot override the configuration of a locked key. Example:

#Save
prgkey5 locked: 1
#Delete
prgkey6 locked:1

When viewing the locked key via the Aastra Web UI, the key is grayed out (disabled) and cannot be changed.

It is also possible to lock parameters in the configuration files, by starting the line with an exclamation mark (!). Example:

! collapsed context user softkey screen:1

11.13.10 Conference key

The hard key for conference in A6730i/31i/39i is defined in the **aastra.cfg** template file, and of course when using MTS to create the configuration file, to send a DTMF digit to the exchange. The possibility to initiate a three part conference locally in the phone is disabled.

The conference softkey sends an xml request to the exchange to initiate a conference.

11.14 Initiating Data from Manager Provisioning

Manager Provisioning (MP) is used to set data e.g. for MNS, TNS, MCT and PEN into the phones from MX-ONE. MP can be used by system administrators and by end-users. The screenshots below shows the menu in MP for setting data on the softkeys in a 6757i terminal.

Figure 15:Manager Provisioning. Key data for 6757i

In the example above the softkeys Log on/off, Services, Corp Dir, Msg Wait, key 9 and More belongs to the first page. Key 10,11, 12, 13, 14 and More to the next page and 15,16,17,18,19 and 20 to the last page.

11.15 Dial Plan

The dial plan is defined via the configuration file. In the **aastra.cfg** file the following parameters are set:

!sip dial plan: "x+^|xx+"*

!sip dial plan terminator: "1"

With this setting the # character will be sent to the PBX in a correct way, even in the middle of a procedure for example **42#B-number#*.

11.16 Authentication code shall not be visible

When entering a service code procedure containing an authorization and PIN code, it is possible to prevent the authorization or PIN code to be stored in the logs. The configuration for this is done in the **aastra.cfg** file. There are two options:

- All the entered digits after the service code are replaced. Example: the user enters *72*99999# where 99999 is the authorization code, *72******# will be shown in the display and in the re-dial list.
- The procedure contains service code + authentication code + number. In this case the function code and number will be shown in as they are entered and the authentication code will be replaced by stars. Example: the user enters *75*99999*6709# where 99999 is the authorization code and 6709 an extension number, *75*******6709# will be shown in the display and in the re-dial list.

The syntax to be used in the aastra.cfg file is shown in following example:

*pin suppression dial plan: *72*(X+)# | *75*(X+)*X+#*

This setting will give the result shown in the example above.

11.17 Free on Second Line

If the telephone shall be able to receive calls on another line although there is a call on line 1 already, Call Waiting must be enabled.

Use the web UI:

Global SIP Settings > Basic SIP Authentication Settings.

Call Waiting is also set per line and this setting overrides the global setting.

The default value is Call Waiting = Enabled.

If the telephone shall send busy when a new call arrives and there is an ongoing call on the first line, Call Waiting shall be set to Disabled.

11.18 Diversion / Call Forward

Pressing the Diversion key gives the the following options:

- **Presence**, see 11.21 Configuring Presence Services on page 56.
- **Follow-me**. The user must enter the number of the new answering position.
- **External follow-me**. The user must enter the external number including the external access code.
- **Do Not Disturb**. When this option is activated the caller will get an extension unavailable message or be forwarded to the answering position, if forwarding is defined by the system administrator. See also 11.19 Do Not Disturb (DND) on page 55.
- **Divert**. The system administrator must define a default personal number list for the extension in MX-ONE, see feature description for Personal Number. When divert is activated from the terminal, the calls to the extension are forwarded to the next position in the personal number list (normally voice mail).

Note: The Divert function for A6700i SIP extensions is a simplified type of diversion and has not the same functionality as the diversion function for digital and analogue telephones.

Note: There is support in Manager Provisioning for setting of the default personal number list, see 11.18.1 Initiate the Divert Settings From Manager Provisioning on page 54.

The Diversion key is set from the PBX and cannot be changed.

The phone internal Call Forward menu is disabled by default in the aastra.cfg file:

call forward disabled: 1

11.18.1 Initiate the Divert Settings From Manager Provisioning

In MX-ONE a default personal number list is used to create the divert function, see feature description for Personal Number.

In Manager Provisioning (MP) a template can be created for initiating an extension with the default personal number list. Using this template the system administrator does not have to fill in the extension number and voice mail number in the personal number list for each extension when new extensions are created. Follow the procedure:

- Manager Telephony System (MTS): Initiate a Common Service Profile (CSP) under the tab *Service Category* with:

  - *Call List Deactivation Forbidden* (which means that the user is not allowed to deactivate the personal number list)

- MP: If a number of new extensions shall be created with default personal number list:
  - Create a new template for an extension with the CSP created above.
  - Select *Personal Number - > Edit*. Use the option ODN (own directory number) as first position in the list. When the template is used, the parameter value ODN will be replaced by the present directory number.
  - In the second position in the personal number list, enter the answering position (normally the voice mail number).
- MP: Create the extension by using the template described above.
- MP: If the default personal number list shall be initiated for an existing extension:
  - Select the extension. Change the CSP to the one for default personal number list. Press *Apply*.
  - Select *Personal Number - > Edit*. Change the phone numbers in the list to the wanted numbers.

11.19 Do Not Disturb (DND)

It is possible to activate individual DND from the Diversion menu in the terminals. The extension must have a certain category to be allowed to activate individual DND. When the feature is activated the forwarding of calls to the extension is dependant on the settings in MX-ONE. No settings in the telephone is necessary for this feature. See also MX-ONE Telephony System Feature List.

It is possible to activate group do not disturb from the telephone with a service code procedure. The extension must have a certain category to be allowed to activate group DND. No settings in the telephone is necessary for this feature. See also MX-ONE Telephony System Feature List.

11.20 Configuring Ring Signals

The adaptation of the ring signals for the market is made from the configuration file. The tables below shows values to be set for Europe / Standard and for North America application systems.

Table 4 Ring signal cadences for Europe / Standard

Ring signal	Parameter in configuration file	Cadence
External	bellcore cadence dr2	350, 300, 350, 5000
Call back	bellcore cadence dr3	300, 400

Table 5 Ring signal cadences for North America

Ring signal	Parameter in configuration file	Cadence
External	bellcore cadence dr2	800, 400, 800, 4000
Call back	bellcore cadence dr3	400, 200, 400, 200, 800, 4000

11.21 Configuring Presence Services

These telephones have menu support for activating of absence reasons (message diversion) under the **Diversion** key. The menus are pushed out from the PBX and no settings are required in the configuration file for the telephone.

When message diversion is active, the lamp at the **Diversion** key is lit.

Note: It is necessary to set the time and date format in MX-ONE for the different absence reasons. Use command:

extension_text with parameter **ics-time-format**

Common answering position(s) must be set as well so the system has a destination number where to redirect traffic when absence reason is active. Use command:

CDCOI with parameters

11.22 Using DNS SRV Resource Records

DNS SRV resource records can be used to implement more alternatives for redundancy than just sip proxy and sip backup proxy. This feature can also be used when a local branch office sip proxy shall be used.

DNS SRV records can be defined in the DNS server or in the *aastra.cfg* file.

The phone performs an SRV lockup when the IP address of the server is a fully qualified domain name (FQDN) and the corresponding port is set to 0.

Here is an example when using *aastra.cfg:*

sip proxy ip: 192.168.0.1
sip proxy port: 5060
sip registrar ip: 0.0.0.0
sip registrar port: 5060

sip backup proxy ip: example.com
sip backup proxy port: 0
sip backup registrar ip: example.com
sip backup registrar port: 0

sip dns srv1 name: _sip_udp.example.com
sip dns srv1 priority: 10
sip dns srv1 weight: 50
sip dns srv1 port: 5060
sip dns srv1 target: mx-one-server2.example.com

sip dns srv2 name: _sip_udp.example.com
sip dns srv2 priority: 10
sip dns srv2 weight: 50
sip dns srv2 port: 5060
sip dns srv2 target: mx-one-server3.example.com

sip dns srv3 name: _sip_udp.example.com
sip dns srv3 priority: 20
sip dns srv3 weight: 60
sip dns srv3 port: 5060
sip dns srv3 target: branch-office-server.example.com

hostfile.txt
192.168.0.2 mx-one-server2.example.com
192.168.0.3 mx-one-server3.example.com
192.168.8.1 branch-office-server.example.com

In the example MX-ONE has ordered the phone to register in MX-ONE server 1 as primary registrar. If server 1 is not available, the phone will use the DNS SRV record with lowest priority, that is record #1 or #2 both with 50 percentage probability, which means that the phone will register towards server 2 or server 3.

MX-ONE must know the IP domain name, that is **example.com** above.

If all MX-ONE servers are unavailable, the phone will use record #3 and register towards the branch office registrar.

The **hostfile.txt** is used to translate the domain name to IP address. This file shall be available on the sw server together with the **aastra.cfg** file. The host file is downloaded to the phone at restart.

When the telephone is registered towards the backup registrar and when the user initiates a call, the INVITE will be sent to the primary registrar first, which causes a delay of the call with 3-4 seconds. The telephone will discover when the primary registrar is working again and register towards this one.

There are some limitations when the telephone is registered towards the backup registrar, see 11.28.1 Limitations on page 60.

11.23 Using the Phone as an Operator Media Device (OMD)

Not applicable.

A telephone is called Operator Media Device when it is used for the speech together with a operator work station application, for example CMG NOW Attendant.

11.24 Central Storage of User Specific Data

The data the that follows the user for the Aastra 6700i family when logging on to another phone is the data stored in MX-ONE, for example MNS, SCA keys and the speed dial keys initiated from MX-ONE. The data locally stored in the phone does not follow the user.

11.25 Configuring the Diffserv Parameter

Diffserv is a model for handling of priority, based on the type of service (TOS) field in the IP packet heading.

The TOS value can be defined in the **aastra.cfg** file and the parameter names are: **tos sip** and **tos rtp.**

The default values are tos sip: 38 AF/(100110) and tos rtp: 46 EF/(101110)

For more information: see Administrator Guide for Aastra Models 6700i and 9000i Series IP SIP Phones.

11.26 Selection of Transport Addresses (Port Numbers)

The table below shows the default port numbers. The ports are possible to change via the **aastra.cfg** file. For more information, see Administrator Guide for Aastra Models 6700i and 9000i Series IP SIP Phones.

Table 6 UDP/TCP default ports used by the phone

Type of signalling	Minimum	Maximum	Comment
RTP	3000		The first media stream, uses 3000. The second mediastream uses 3002 and so on.
RTCP	3000+1		RTP port + 1
SIP	5060	5060	
SIP secure port	5061	5061	When using the phone in SIP / TLS mode.
MX-ONE Aastra XML API port	22222	22222	Used in combination with http:// when configuring /Logon and /Startup
MX-ONE Aastra XML API secure port	22223	22223	Used in combination with https:// when configuring /Logon and /Startup
Configuration Server HTTP	80	80	when downloading configuration and firmware files
Configuration Server HTTPS	443	443	when downloading configuration and firmware files

11.27 Registration Distribution

When the registration distribution feature is enabled in the system, the Aastra 6700i phones will be configured to register in the server with the generic extension data.

In the MX-ONE concept called HLR (Home Location Register) server, the configuration is a part of the initial REGISTER procedure. The phones will as the main rule be registered in their home server, but if the

HLR server has reached its limit, an alternative server will be able to accept the registration.

When this feature is used, the balancing of the registration load has to be considered already during the deployment of the system, e.g. an even initiation of the generic extensions among the available servers.

To activate the registration distribution feature in MX-ONE enter the command:

extension_registration_distribution -i

11.28 Redundancy

The primary proxy IP address shall be set according to 7.3.1 Log on with Soft key on page 21. It is also possible to define in the **aastra.cfg** file a redundant IP address with the parameters:
sip backup proxy ip
sip backup registrar ip

The IP address to set shall be equal in both of these parameters. The redundant ip address will be used by the telephone if the sip proxy server with the primary ip address does not respond.

When the telephone is registered towards the backup sip registrar, the telephone tries to register towards the primary server. If this server responds, the phone tries to register towards the primary proxy server again.

There is support for redundancy settings in Manager TS.

For more information about the redundancy functionality in MX-ONE, see feature description for HLR Redundancy.

11.28.1 Limitations

When the phone loses contact with the primary SIP registrar, it can take up to 10 minutes (refresh of the registration time) before the telephone registers towards the backup registrar.

Aastra 6700i terminals, deployed with Log On/Off xml key, cannot use the key when the primary proxy/registrar is not responding. As a work-around it is possible to log on with the free seating procedure (*11*PIN*extension number#).

If the terminal is registered towards the backup registrar and if the user logs off with the #11# procedure, the IP address to the primary SIP proxy is lost and it is only possible to log on towards the backup SIP proxy. To go back to primary SIP proxy, the local configuration settings have to be removed in the phone, see 9.2 Remove local configuration settings on page 27.

When the telephone is registered towards the backup registrar and when the user initiates a call, the INVITE will be sent to the primary proxy first and then to the backup proxy, which causes a delay of the call with 3-4 seconds.

11.29 Registration at Branch Offices

The branch office scenario means that the telephones are registered to to PBX in the main office and if the connection to the main office fails, the phones shall register to a local SIP server.

When the connection to the main office is working again, the telephones shall register towards this PBX again.

The way to configure this is to use *sip backup proxy ip* and *sip backup registrar ip* in the **aastra.cfg** file, see section 11.28 Redundancy on page 60 or see 11.22 Using DNS SRV Resource Records on page 57.

11.30 Voice Mail

When a user has got a voice mail and the message waiting key is flashing, the user can listen to his voice mail by pressing this key. The telephone will send the *32# procedure to the system.

If there is no message waiting, it can still be useful to call the voice mail system. This is done by pressing the Services key and select VoiceMail. In this case the phone will use the directory number to access the voice mail system.

The settings for these options are defined in the aastra.cfg file. Below is an example:

*sip vmail:"*32#"*
*sip line1 vmail:"*32#"*
sip explicit mwi subscription: 1
servicesscript: http://$$PROXYURL$$:22222/services?user=$$SIPUSERNAME$$
voicemailnr=12345

11.31 Corporate Directory

From the phone it is possible to search in a corporate directory via the XML interface described in XML API for Aastra SIP phones.

From CMG7.5-SP1 the XML support for searching in the corporate directory is included.

To be able to access the directory function some parameters in the **aastra.cfg** file of the phone have to be set, see the example below:

softkey6 label: "Corp Dir"
softkey6 type: xml
softkey6 value: http://<CMG server>/xml/directory/CorpDir.php
softkey6 line: global
softkey6 states: idle, connected, incoming, outgoing

The phone sends a http request with the search criteria to the directory server and receives a list with the search result. The answer is in xml format.

The user can select the phone number in the search result and initiate a call.

11.31.1 CMG directory

For details how to set up CMG to access the corporate directory from the 6700i phones, see Corporate Directory for IP phone, Installation & Configuration Guide in the CMG CPI library.

11.32 Call Park Pool

For a detailed description of the Call Park Pool feature in an MX-ONE environment, see operational directions for Call Park Pool.

No configuration in the phone is needed for this feature.

11.33 Intercom

It is possible to define Intercom functionality on one or several programmable keys. When pressing the Intercom key, the telephone initiates a call towards the other predefined party and the call is automatically answered.

For a detailed description of the Intercom feature in an MX-ONE environment, see operational directions for Intercom.

No configuration in the phone is needed for this feature.

12 Passwords and PIN Codes

The following passwords or PIN codes are used when working with these phones:

- PIN code for registering the phones to MX-ONE. The user can change the PIN code with the procedure: *74*old PIN*new PIN#

  It is recommended to use PIN code to avoid that an end-user can log on with another end-user’s directory number.
- Administrator password for accessing the phone using the phones' web interface or the phone menus.
- User password for accessing the phone using the phone's web interface or phone menus.

12.1 Changing the Administrator Password

The administrator password can be changed from the aastra.cfg file, see Administrator Guide for Aastra Models 6700i and 9000i Series IP SIP Phones. The password can only consist of digits

The user name is: **admin**

The default password is **22222**.

12.2 Web Interface Passwords for End Users

End users can use a Web browser to access the phone's Web interface. This interface can be used when working with contacts and other user specific data.

The user name is: **user**

There is no password by default.

The user password can be initiated/changed via the phone UI:

Options > **Option List** > **User Password**

The user password can also be changed via the web UI.

Click on: **Operations** > **User Password**

13 Headset

Wireless headsets according to the DHSG protocol as well as headsets with cable can be used with 6735i/37i, 6737/57i, 6739i and 6753i.

To connect the phone to a DHSG compatible cordless headset, a special cable from Aastra must be used. The article number is: 62-001134-00.

To enable the headset port via the phone UI:

Options > **Preferences** > **Set Audio** > **Audio Mode**

14 Expansion Modules

There are two types of expansion modules; M670i and M675i.

M670i is a key panel unit with 36 keys. Paper labels are used to label the keys. Maximum 3 modules can be connected to the telephone. The following models has support for the key panel unit: 6753i, 6735i/55i, 6737i/57i and 6739i.

M675i is a display panel unit with 60 soft keys, with 20 softkeys on each page which means 3 pages. A LCD display is used to label the keys. Maximum 3 modules can be connected to the telephone. The following models has support for the key panel unit: 6735i/55i, 6737i/57i and 6739i.

For installation of the modules, see the phone specific Installation Guide.

When adding an expansion module on the telephone, the keys for MNS, TNS (defined in the PBX), MCT and PEN are moved from the telephone to the expansion module.

If shortcut keys shall exist on both the phone and on the expansion module, the key numbers that do not fit on the expansion module will ‘overflow’ to the telephone display. Key number 1 to 60 are placed on the expansion module and key 61 and upwards on the telephone, in the case when one module is used.

15 Emergency Calls

Even if the telephone is not registered to the PBX, it is possible to make emergency calls. The *sip proxy ip* parameter in the configuration file, defines where the telephone sends the INVITE with the SOS number. No registrar is required for this.

For more information how to set up the sent A-number, see Emergency Calls, SOS calls (5/15431-ANF90143) in the CPI library.

16 VoIP Recording

It is possible to record voice calls to a central recording equipment. The phones that shall have recording are monitored via the CSTA interface and this means that an Application Link or an Open Application Server (OAS) must be used to provide the CTI interface to the recording system. The call events and the IP address to the phones to be monitored are sent over the CSTA interface.

For more information about the recording solution for MX-ONE Telephony Server see *Description for Voice Recording* and the *Interface Description for VoIP Recording Interface.*

The signalling between the recording system and the IP phones is based on SIP. The recording system sends an INVITE message to the phone to inform about the IP address to where the voice packets shall be sent. A SIP ACK message orders the phone to start forwarding the received and transmitted RTP streams to the logger.

There are the following options

1. **Total recording**: the recording system orders the telephone to start the recording dependent on the recording policy. All calls or only external calls to the monitored extensions are recorded for example.
2. **Record on demand**: the user can start and stop the recording by pressing the recording key.

Note: It is only possible to record IP phones. No other types of phones shall be monitored.

The voice stream is sent un-encrypted to the recording equipment, if the original call is without encryption. If the call is encrypted, the telephone forwards an encrypted voice stream to the recorder. In this case the encryption keys are sent via the CSTA interface to the recording equipment.

The icon for recording in the display, is shown when the telephone forwards the RTP stream to the recording system. Moreover it is possible to configure the phone to play a periodic beep tone.

16.1 Configuration at Total Recording

The recommendation is to use dynamic recording sessions (i.e. per call), although it is also possible to use static (i.e. per the duration that the phone is registered). The type of recording session is configured in the recording system.

Below is an example how to set the parameters in the configuration file:

recorder address1: 192.168.1.20
recorder address2: 192.168.1.21
#recording destination1:
recording destination beep: 0
sip services transport protocol: 1
sip services port: 7300

Recorder addressN specifies the trusted IP addresses corresponding to the recording system.

Recording destinationN specifies the trusted IP addresses corresponding to the destination where the RTP/SRTP streams should be sent. If these parameters are left undefined, no authentication checks are performed. This is the case in this example.

Recording destination beep defines if a beep shall be heard to notify that recording is ongoing.

Sip services transport protocol. The following options exist:

- **-1**: (default), which means that the same protocol will be used as defined in the parameter *sip transport protocol* parameter.
- **0**: TCP/UDP
- **1**: UDP
- **2**: TCP

16.2 Configuration at Recording on Demand

The prerequisite is that the settings for active recording is done, see section 16.1 Configuration at Total Recording on page 67.

When the user has pressed the recording key an icon is shown in the display when the phone has got a confirmation from the recording system that the recording has started.

The shortcut key for recording is initiated from the PBX.

The URL to the recording system that the telephone sends when the user presses the recording key, can be defined in one of the following ways:

- MX-ONE command **extension_key**. The advantage is that in a free seating environment, the recording key with the associated url will follow the user.
- In the configuration file for the phone. Can be used where it is not necessary that the key follows the user.

In the second option, the recording key is defined as a key of the type **xml** with the url as parameter value. For example with the recording system from the vendor ASC:

- **Start recording:**
  http://192.105.88.152:8080/XVOIPService?page=START&OPN=$SIPUSERNAME$$

  where n is the extension number added by the telephone. The IP address and port number must match the recorder's listening IP address and port number.

- **Stop recording:**
  http://192.105.88.152:8080/XVOIPService?page=STOP&OPN=$SIPUSERNAME$$

If the URL contains any characters considered as special by Aastra 6700i, they must be escape encoded, see table below.

Table 7 Escape encoding

Character	Symbol	Escape Sequence
Ampersand	&	&
Quote	“	"
Apostrophe	‘	'
Left angle bracket	<	<
Right angle bracket	>	>

17 Quality of Service (QoS)

It is not possible to view the QoS statistics via MX-ONE.

18 DHCP server

18.1 Data from DHCP

The phone has support for DHCP by which the following IP configuration data can be provided:

- Own IP address, subnet mask and default gateway, received in the DHCP standard fields (1 and 3).
- The VLAN used for the phone can generally be set in option 132 or be part of Option 43. If the phone's configuration has another value than that of the option value it will configure according to the Option 132 value and making a reboot.
- IP address to the software server. The path to the firmware to be downloaded from the software server can also be provided as well as the protocol to be used. The recommendation is to use DHCP option 66 (TFTP server name), but DHCP option 60 (vendor class identifier) and option 43 (vendor specific information field) can also be used.

The following examples show the different possibilities on how to use option 66,160 or 159 in order to get the IP address or host and its path to the software server. For http and https it is possible to define the port. Default port for http is 80 and default port for https is 443:

http://192.168.1.45

http://192.168.1.45/path

http://192.168.1.45:8080/path

http://srv.example.com/path

The default dhcp precedence order is 43, 160, 159, 66. So if the phone receives the software server configuration in both option 66 and option 43, then option 43 takes precedence over option 66.

If option 66 is already in use, it is possible to set the configuration server in either option 160 or 159 instead.

18.2 DHCP Settings for Option 66

Enter the URL to the software server according to the example in 18.1 Data from DHCP on page 71.

18.3 DHCP Settings for Option 43 and 60

DHCP option 60 (vendor class identifier) and option 43 (vendor specific information field) can also be used to get the software server address and also to load a unique configuration file dependant on telephone type.

The first step is to initiate option 60 for each telephone type:

Table 8 Identifier values to be defined in option 60

Model	Identifier Value
6730i	AastraIPPhone6730i
6731i	AastraIPPhone6731i
6735i	AastraIPPhone6735i
6737i	AastraIPPhone6737i
6739i	AastraIPPhone6739i
6753i	AastraIPPhone53i
6755i	AastraIPPhone55i
6757i	AastraIPPhone57i

After option 60 has been entered into the DHCP server, the data in option 43 has to be entered. The following options exist:

Table 9 Options that can be set in option 43.

Code	Description
02	Configuration server (protocol, server and path). Syntax: string
03	RCS. Not used in a MX-ONE environment
08	Header to activate the VLAN transfer to the phone. Syntax: 16 bytes character string “Aastra(space)Telecom(space)(space)” i. e. 41617374726120546 56c65636f6d2020
09	VLAN identity (1-4094) Syntax: 4 bytes whereas the first and second byte must be 0x00 and third and fourth byte the VLAN id. Example: 100 in decimal is 00 00 00 64 in hex.

For an example how to configure option 60 and 43 in a Linux environment, see the Administrator Guide for Aastra IP SIP Phones

Below is an example showing how to configure DHCP in a Windows environment.

18.3.1 Define Vendor Class

Select Define Vendor Class in the drop down list.

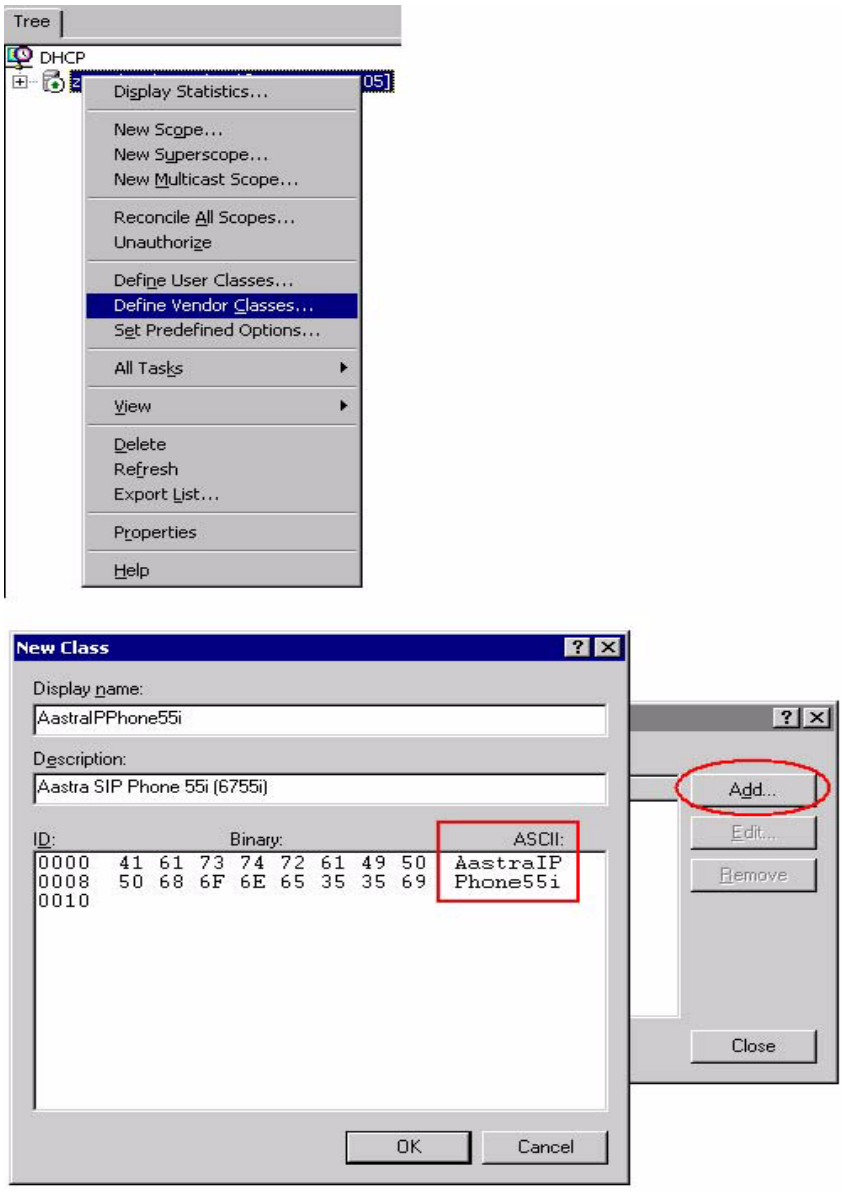


Figure 16:Define and add the vendor class

To enter the Vendor Class ID, click on the right side below **ASCII** in the large form field. Enter the Identifier Value from table 7 above.

Repeat this step for each phone model that should be served by this DHCP server.

18.3.2 Set Predefined Options

Select Set Predefined Options to get the menu to enter the option 43 data.

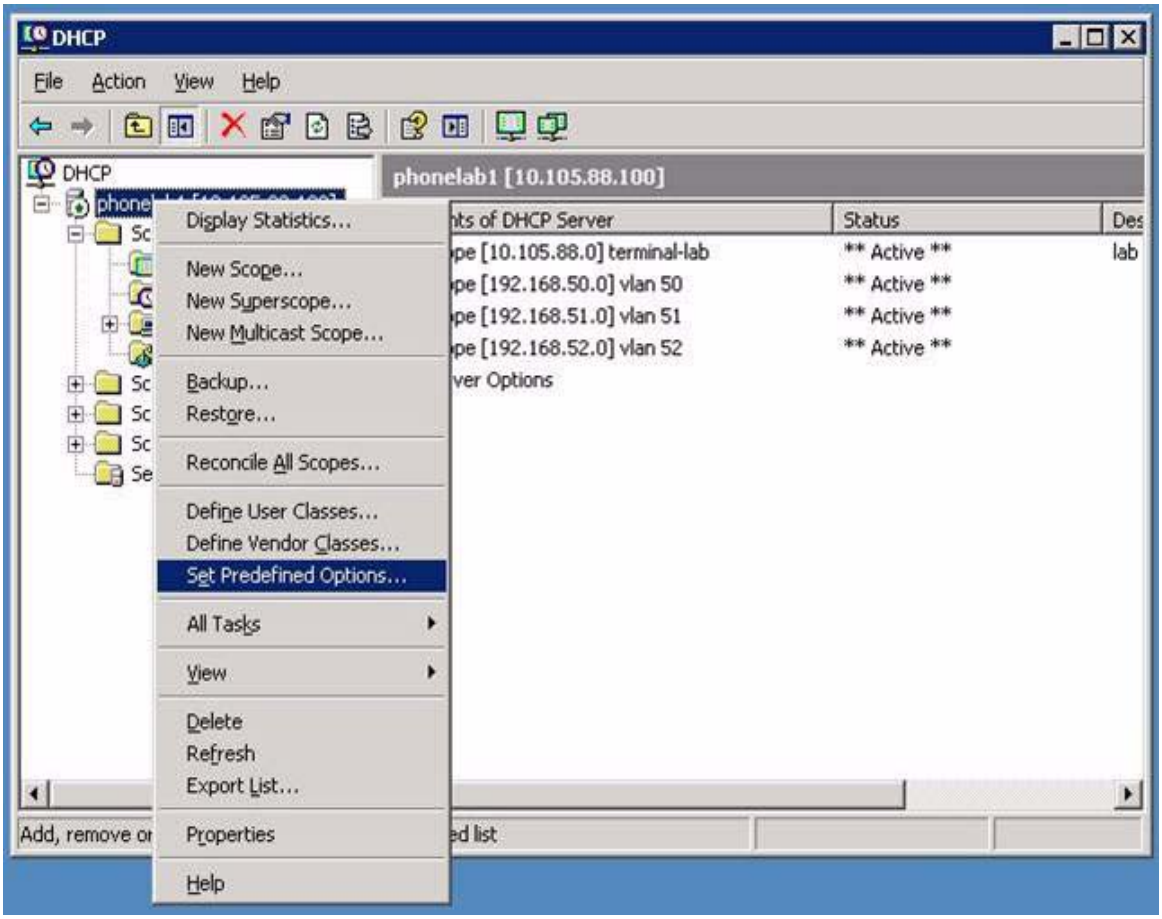


Figure 17:Set Predefined Options

Select appropriate option class from the drop down list and press the **Add** button.

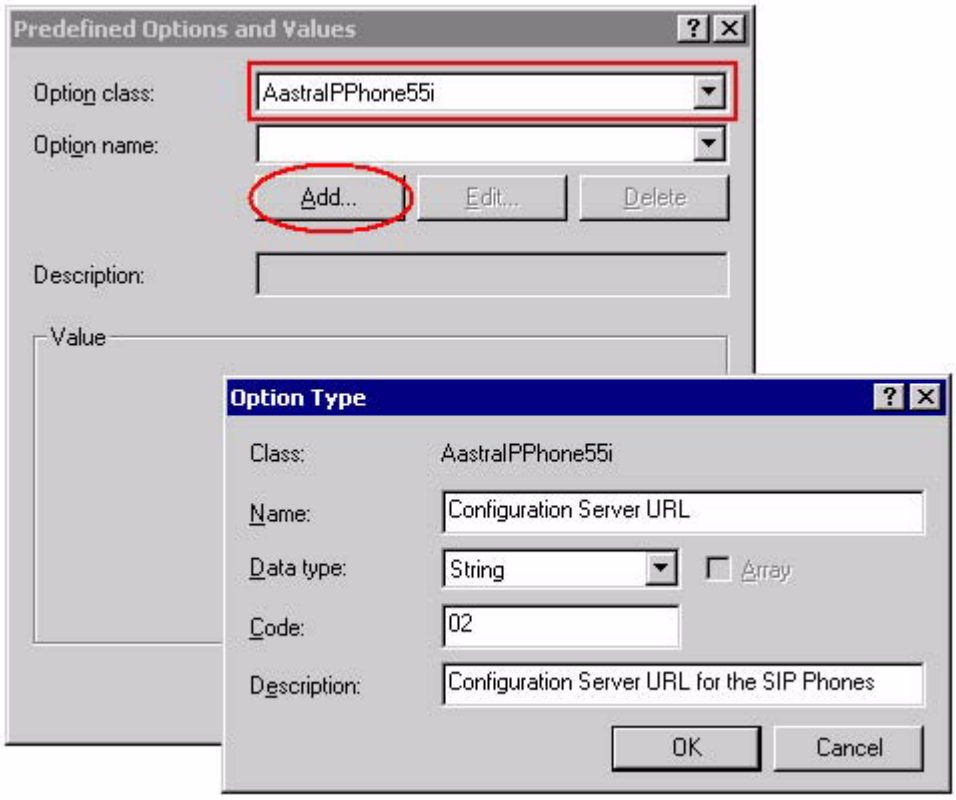


Figure 18:Predefined Options and Values

The data in the Option Type menu has to be entered manually:

Name: Configuration Server URL

Data type: String

Code: 02

Repeat this for each phone model that should be served by this DHCP server.

If VLAN identity shall be provided via option 43, repeat this for code 08 and code 09, see table 9 Options that can be set in option 43. on page 72

18.3.3 Set Scope Options

The last step is to set the URL string.

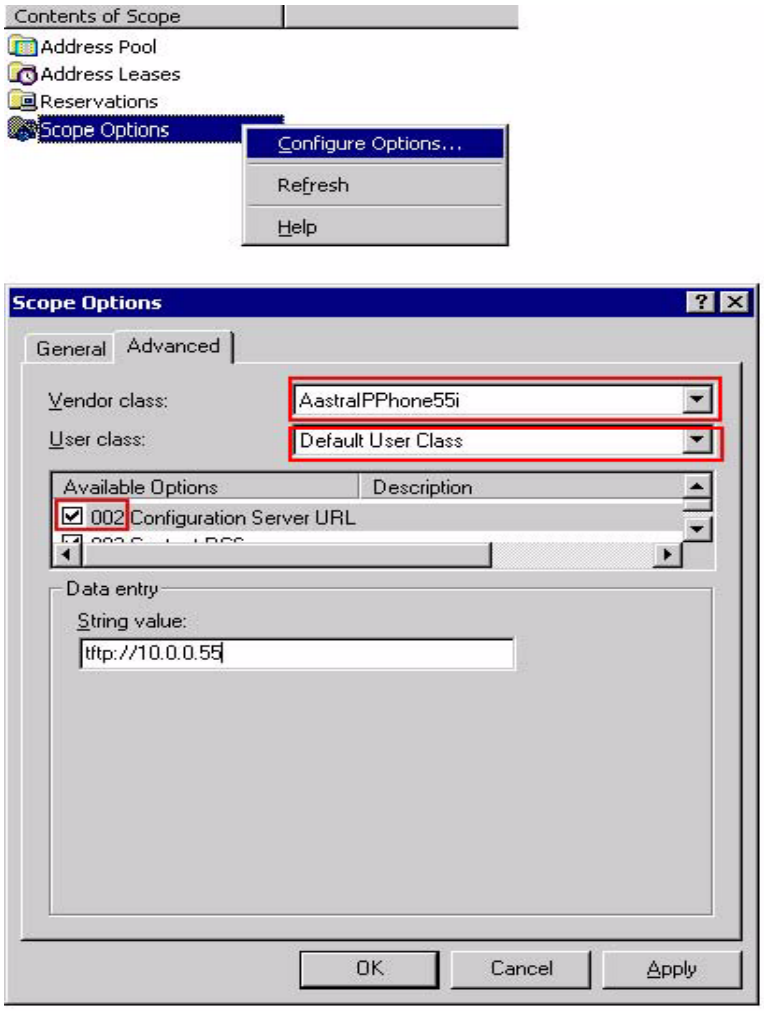


Figure 19:Set Scope Options

Select appropriate Vendor class and set the User class to *Default User Class*. Activate option 002 and enter the URL of the software server (configuration server) in the input field String value.

Repeat this for each phone model that should be served by this DHCP server.

If VLAN identity shall be provided via option 43, repeat this for code 08 and code 09, see table 9 Options that can be set in option 43. on page 72

19 Security

This section describes the encrypted configuration files, SIP signaling with TLS and media with SRTP.

19.1 Encrypted Configuration Files

The **aasta.cfg, <model>.cfg** and <**MAC>.cfg** files can be encrypted and downloaded to the phone from the software server with the http or https protocol. Aastra provides a tool for Windows and Linux to encrypt the configuration files. This tool is called **anacrypt**. Use the following procedure:

1. Create the file **security.tuz** with the encrypted site key:

   anacrypt -i -p <shared_password>

2. Encrypt the **aastra.cfg** file:

   anacrypt aastra.cfg -p <shared_password>

3. Encrypt the **<model>.cfg** file:

   anacrypt <model>.cfg -p <shared_password>

4. If MAC configuration files are used, encrypt the **<MAC>.cfg** file:

   anacrypt <mac>.cfg -m -p <shared_password>

   To encrypt all MAC configuration files in a directory:

   anacrypt <mac>.cfg -d <dir> -m -p <shared_password>

5. Store **security.tuz**, **aastra.tuz** and **<mac>.tuz** on the software server. Reboot the telephones.

The shared password can be 4-32 alphanumeric characters.

The anacrypt tool can be downloaded from **www.aastra.com**.

19.2 TLS

IP Phones support a transport protocol called **Transport Layer Security (TLS)** . TLS is a protocol that ensures communication privacy between the SIP phones and the Internet. TLS ensures that no third party may eavesdrop or tamper with any message. persistent TLS is the only mode supported by MX-ONE. If Ingate is the access for the phone both persistent mutual TLS and persistent TLS are supported. persistent TLS

means that the phone will setup a TLS session which it will keep as long as it is registered (logged on). Both the server and the phone will make use of the session to setup calls. Persistent mutual TLS is referering to the additional mutuality in the TLS handshake where the server requests the the client's signed certificate. Otherwise only the client requests the servers certificate.

19.3 SRTP

The IP Phones include support for Secure Real-time Transfer Protocol (SRTP), using Session Description Protocol Security (SDES) key negotiation, for encryption and authentication of RTP/RTCP messages sent and received by the Aastra IP phones on your network.

The administrator can choose among the following options:

- SRTP Disabled (default): IP phone generates and receives non secured RTP calls. If the IP phone gets a call from a SRTP enabled phone, it ignores SRTP and tries to answer the call using RTP. If the receiving phone has *SRTP only* enabled, the call fails; however, if it has SRTP preferred enabled, it will accept RTP calls.
- SRTP Preferred: IP phone generates RTP secured calls, and accepts both secured and non-secured RTP calls. If the receiving phone is not SRTP enabled, it sends non-secured RTP calls instead.
- SRTP Only: IP phone generates and accepts SRTP secured calls only; all other calls are rejected (fail)

  SRTP Only: IP phone generates and accepts SRTP secured calls only; all other calls are rejected (fail).

19.4 How to enable security on 6700i terminals and MX-ONE

A number measures have to be done in MX-ONE and in the configuration file in the phone.

There is support in MTS for enabling security in MX-ONE and in the 6700i phones.

The steps to enable security are:

1. MX-ONE: For setup of security and security policy, see operational directions VoIP Security (82/15431-ANF90114) in the CPI library.

2. MX-ONE: For certificate handling see operational directions Certificate Management (132/15431-ANF90114) in the CPI library.

3. 6700i phones: The only certificate that is necessary is the root certificate. The key storage for MX-ONE certificates is **/etc/opt/eri_sn/certs/**. The root CA is called, CA.pem. Copy CA.pem to the configuration server, i.e. the same directory as where aastra.cfg resides. You may set the file name of the root certificate via MTS or directly in the **aastra.cfg**.

4. Phone **aastra.cfg** file: below is an example of the parameters:

```
sips persistent tls: 1
sip outbound support: 1
sip transport protocol: 4 # 0=UDP&TCP,1=UDP,2=TCP,4=TLS
sips trusted certificates: CA.pem
sip outbound proxy ip: 192.168.110.20
sip outbound proxy port: 5061
sip backup outbound proxy:192.168.5.3
sip backup outbound proxy port:5061

sip srtp mode: 1 #0-RTP,1-SRTP preffered,2-SRTP only
```

With the backup outbound parameters security is enabled towards the backup server.

If the XML keys provisioned by MX-ONE shall use https, the following setting is required. port 22223 will trigger MX-ONE to provision XML keys for Logon/Logoff and Diversion as “https” and port 22223, which is the TLS port for Aastra-XML.

```
action uri startup:
”https://192.168.110.20:22223/Startup?user=$$SIPUSER-
NAME$$”

services script:
https://192.168.110.20:22223/Services?user=$$SIPUSER-
NAME$$&voicemailnr=<voice mail number>

#download protocol HTTP,HTTPS,FTP,TFTP
download protocol:HTTPS
https server:<IP address of Configuration Server (Tomcat or
webserver of choice)>
https port:443 #443 is the standardport for https
https path:aastra67xxi //path on Tomcat
https client method:"TLS 1.0"
https user certificates:CA.pem #root CA
```

Phone **<model>.cfg**:
Either the Logon keys are removed. Then you rely on that the users logon when prompted due to reboot (triggered by action uri startup in aastra.cfg), or the /Logon key value needs to be set to *“https://192.168.110.20:22223/Logon?user=$$SIPUSERNAME$$”*.

Check the CMG documentation if Corporate directory is to use https (TLS), for example:

```
https://192.20.22.11/xml/directory/CorpDir.php
```

5. Per default a time server (using NTP as protocol) needs to be enabled via DHCP Option 42 or via configuration parameters. The configuration parameter has precedence over Option 42. The phone must have a valid date and time in order to verify the server certificate's expiry time. As TLS is a per-hop protocol. It is the server certificate of 'outbound proxy ip' which is verified. In this example this would be an MX-ONE server.

   Configuration parameters

   ```
   time server disabled:0 #0-enabled,1-disabled
   time server1:<ip address or host>
   ```

   #ref: http://www.pool.ntp.org/en/use.html

6. If no NTP servers are accessible for some reason it is possible to disable the check for expire date via WebGUI(Network) or configuration parameter. This will also have the effect that there is no date and time indication on the phone.

   ```
   https validate expires: 0 #0-disabled, 1-enabled
   ```

19.5 How to enable security for home worker on Aastra 6700i

If Ingate is used as Session Border Controller (SBC), follow the Installation Guide *How to Install an Ingate Solution for Aastra Teleworker Solutions in Stand-alone mode or DMZ/LAN mode behind existing Firewall* in the CPI library.

The principle used here is to configure the SBC to have secure communication on the outside towards the home worker Aastra 6700i terminal and unsecure communication on the inside towards MX-ONE.

The TLS setup described here will be persistent TLS. If your deployment requires an even more secure setup, 'persistent mutual TLS', then also read the Appendix, "Teleworker with persistent mutual TLS".

Furthermore the assumption is that the user would like to be able to use the terminal in the office and to bring the terminal home (home worker). For this reason two configuration server directories are set up, inOffice accessible via http and atHome accessible via https.

The only setting required by the end user is to change the Configuration Server via phoneUI: **Options** > **Admin Menu** > [6739i; **Advanced**] > **Cfg. Svr.**, choose HTTP or HTTPS in the Download Protocol list. Activate setting by requesting **Options** > **Restart**.

The benefit having the SBC server certificate signed by a commercial CA (Verisign, Thawte, GeoTrust, Comodo or CyberTrust) is that these root CAs are pre loaded in the phone firmware. A root CA is required prior to the TLS handshake with the Configuration Server when HTTPS is used as download protocol.

The following example shows how to get it working with an SBC that has a self-signed server certificate (meaning that the root CA is the server who has the server certificate). The drawback is that the phone needs to boot up in the office before it can be brought home in order to load the root CA, which is used when the phone boots up and access the configuration server via https at home. However, the phone will loose the loaded CA on “Factory Reset” or if a new firmware is found in the configuration server.

1. Setup a webserver like Apache and create the path matching the configuration server setting in the phone configuration. If Apache is used the /var/www/html/ is the root for the path set in the phone. So here you create the directories inOffice/ and atHome/.

2. The InOffice directory shall consist of model specific configuration files, aastra.cfg and the phone FW (see above). Note, that the root certificates are loaded but not used as the setting is TCP for SIP and RTP for media.

   Phone **aastra.cfg** file:
   #Only changes from the aastra template is described

   action uri startup:
   ”http://$$PROXYURL$$:22222/Startup?user=$$SIPUSERNAME$$”

   services script: https://$$PROXYURL$$:22222/Services?user=$$SIPUSER-NAME$$&voicemailnr=<voice mail number>

   #download protocol HTTP,HTTPS,FTP,TFTP
   download protocol:HTTP
   http server:<webserver IP address>
   http port:80
   http path:inOffice
   https server:<SBC outside IP address>
   https port:444 #SBC TLS port relay to webserver
   https path:atHome

   https client method:"TLS 1.0"
   https user certificates:CA.pem #root CA

   sip transport protocol: 1 #1-UDP,2=TCP,4=TLS
   sips trusted certificates: CA.pem #root CA

   sip srtp mode: 0 #0-RTP,2-SRTP only

   sip proxy ip: 192.168.110.20
   sip proxy port: 5060
   sip registrar ip: 0.0.0.0
   sip registrar port: 5060

time server disabled:0 #0-NTP enabled
time server1:<NTP server> #skip this setting if DHCP Options 42 is used

3. The atHome directory shall consist of model specific configuration files, aastra.cfg and if you have a selfsigned certificate you should skip the phone FW as an upgrade will remove the certificate loaded.

   Only changes from the aastra template is described. Set “https” and the secure port “22223” to invoke XML Requests over TLS

   Assuming SBC outside IP address to be: `193.10.10.10`

   Phone **<model>.cfg**:
   Either the Logon keys are removed. Then you rely on that the users logon when prompted due to reboot (triggered by action uri startup in aastra.cfg), or the /Logon key value needs to be set to *“https://193.10.10.10:22223/Logon?user=$$SIPUSERNAME$$”.*
   If Corporate directory is to be used a TLS port relay can be configured in Ingate the same way as port 444 is setup towards the configuration server), Let’s say port 445 is set up then the Corporate Directory key value would be:
   “https://193.10.10.10:445/xml/directory/CorpDir.php”

   Phone **aastra.cfg** file:

   action uri startup: ”https://193.10.10.10:22223/Startup?user=$$SIPUSERNAME$$”

   services script: https://193.10.10.10:22223/Services?user=$$SIPUSERNAME$$&voicemailnr=<voice mail number>

   #download protocol HTTP,HTTPS,FTP,TFTP
   download protocol:HTTPS
   http server:<webserver IP address>
   http port:80
   http path:inOffice
   https server:<SBC outside IP address>
   https port:444 #SBC TLS port relay to webserver
   https path:atHome
   https client method:"TLS 1.0"
   https user certificates:CA.pem #root CA

   sips persistent tls:1
   sip outbound support: 1
   sip transport protocol: 4 #1-UDP,2=TCP,4=TLS
   sips trusted certificates: CA.pem #root CA
   sip outbound proxy:193.10.10.10
   sip outbound proxy port:5061
   sip srtp mode: 2 #0-RTP,2-SRTP only
   The proxy and registrar is set via /Startup or /Logon, which will be the MX-ONE server receiving the XML Request according to the

Ingate Relay setting for port 22223. However, if extension_registration_distribution is active the proxy, registrar will be set according to the extension's Home Location Register (HLR) (see the lim setting in command extension -p)

sip proxy ip: 0.0.0.0
sip proxy port: 0
sip registrar ip: 0.0.0.0
sip registrar port: 0

time server disabled:0 #0-NTP enabled
#skip the server setting below if DHCP Options 42 is used
time server1:<NTP server>

Make sure the NTP server is accessible from the home network. You may use a server from http://www.pool.ntp.org/en/, as for example 0.se.pool.ntp.org

20 Troubleshooting

20.1 Capture logfiles via SYSLOG

When log files for troubleshooting purpose shall be retrieved from the telephone, it is possible to use the external syslog feature in Linux for storing or the Kiwi Syslog Server.

Setup the SYSLOGD server

In the MX-ONE system: Uncomment the following line in
/etc/syslog-ng/syslog-ng.conf.in
#
uncomment to process log messages from network:
#
udp(ip("0.0.0.0") port(514));

Run 'SuSEconfig' to initiate the changes to the syslog-ng configuration

Restart the syslog process:
/etc/init.d/syslog restart

Verify that syslog is listening on port 514:
linux-jloz:~ # netstat -nap | grep 514
udp 0 0 0.0.0.0:514 0.0.0.0:* 8043/syslog-ng

Setup the Kiwi Syslog Server

This is a syslog server for Windows. There is a free version that can be downloaded from http://www.kiwisyslog.com.

Go to **File** > **Setup** and set the UDP listen port. This must match the port which has been set in the phone. The default syslog port is 514.

Setup In the Terminal

Use the WebUI:

Advanced Settings > Troubleshooting > Log IP / Log Port

Enter the IP address and port number (514) to the syslog server where the log shall be stored.

Enter the debug levels according to the table below, into the web UI:

Table 10 Debug level

Debug level	Value
Fatal errors	1 (default)
Errors	2
Warnings	4
Init	8
Functions	16
Info	32
All debug levels off	0
All debug levels on	65535

The debug levels can be combined. Example: Fatal errors + Errors + Warnings = 1 + 2 + 4 = 7.

When fault reporting in TeamTrack, the traces shall normally be with the highest debug level.

Use the web UI to save the log files:

Advanced Settings > Troubleshooting > Support information

The following log files are available: **local.cfg**, **server.cfg** and **crash.log**. It is also possible to view the **Task and Stack Status.**

For more information about troubleshooting, see Administrator Guide for Aastra Models 6700i and 9000i Series IP SIP Phones.

20.2 Issues with DHCP Options

If there is a conflict in the network on what the DHCP Options are used for, you can change or turn off the use of DHCP Options locally on each phone.

On the phone press **Options key** > **Advanced** > **Network** > **DHCP Settings** > **DHCP Download Options**. The possible values alternative values are to ignore any dhcp options ”Disabled” or to set which dhcp option to listen to. It is also possible to change the DHCP options via the WebUI.

In order to keep this setting after the admin has run the MX-ONE command “extension_unregistration --forced “, which will clear local settings in the phone, it makes sense to have the same setting in the aastra.cfg file, parameter.

a) *dhcp config option override: [-1(Disabled),0(Default),43,66,159,160]*

21 Appendix

21.1 Teleworker with persistent mutual TLS (MTLS)

Reference http://en.wikipedia.org/wiki/Transport_Layer_Security

Any TLS will encrypt the SIP signaling to prevent eavesdropping. However if the simple TLS handshake used in ‘persistent TLS’ is used only the server is authenticated by its certificate (this is the method used in chapter 19.5 “How to enable security for home worker on Aastra 6700i”). In a client-authenticated TLS handshake (also referred to as mutual TLS), the server will request to authenticate the client based on its certificate as well. In ‘Persistent mutual TLS’ the client will make a client-authenticated TLS handshake and the TLS session is kept by the client as long as the phone is registered (logged on).

Why would you do the effort to create both server certificate and client certificate? The SBC who is the access point for traffic from a teleworker (perhaps working from home) and is configured to do ‘client-authenticated TLS’ will only allow clients (phones) which offers the expected client certificate in the handshake. So this is a way to block unwanted registration attempts early. If a registration reaches the MX-ONE, the only check would be to require a password for the registering directory number, which is recommended anyway. Also check the SBC manual for other ways to block/allow traffic.

21.1.1 Create persistent MTLS using a Enterprise CA (openssl) to sign both server and client certificate and configure the SBC

Prerequisites using openssl on the linux server acting as Enterprise CA. In this example you will sign certificates. Be careful with the root password to this server as the CAcan sign any TLS Request. This chapter will show how to sign certificates.

In a shell do the following. (Note that this is an example. Use your own passwords).
As root do:

```
>cd /etc/pki (or wherever the certs should live)
>mkdir sbc
>cd sbc
>mkdir private
>chmod 0700 private
>echo “01” > serial
```

Create a CA
```
>openssl genrsa -aes256 -out private/cakey.pem 1024
```
[password: test]
```
>openssl req -new -x509 -days 3650 -key private/cakey.pem
-out ca.pem -set_serial 1
```
[answer cert questions accordingly]

Keep the shell open. We will use it for openssl commands from time to time in this chapter. Now, generate the TLS cert request on the SBC, which shall be signed by CA on openssl to be the Server certificate, when imported back to the SBC.

Logon to Ingate as Admin via webbrowser (if you open the browser on the linux machine it is easier to download and upload files later) and go to
Basic Configuration->Certificates->Private Certificates: <Create New>
- Name: TLS-CA_SIGNED
- CN: <public IP address of the SBC> --> Create an X.509 certificate request

Download the output, certreq.req, to etc/pki/sbc
Sign the TLS request using the CA
```
>openssl x509 -req -in certreq.req -out sbccert.pem -CAkey
private/cakey.pem -days 3650 -CAcreateserial -CAserial
ca.seq
```
Output: signed server certificate, sbccert.pem

On Ingate web, import the signed server certificate. Go to
Basic Configuration->Certificates-> <import> and upload “sbccert.pem” --> Ingate should show “certificate has been imported” This certifiate shall now be used in the rules on what to authenticate to. Go to SIP Services->Signaling Encryption: TLS CA Certificates.

Enable Client Certificate Check on SBC (mutual Authentication). Go to SIP Services->Signaling Encryption:TLS Connections On Different IP Address,
IP: outside (IP equal to CN in sbccert.pem),
Own Certificate: the label name for sbccert.pem
Use CN FQDN:No
Require Client Cert:Yes
Accept Methods: TLSv1

Generate a phone certificate (client certificate) and sign it by CA.First TLS cert request is created and then it is signed by the CA

```
>openssl req -new -newkey rsa:1024 -out phone_csr.pem
-nodes -keyout private/phonekey.pem -days 3650
```
[answer cert questions accordingly]CN: Aastra IP Phone

```
>openssl x509 -req -in phone_csr.pem -out phonecert.pem -CA
ca.pem -CAkeyprivate/cakey.pem -days 3650 -CAcreateserial
-CAserial ca.seq
```

21.1.2 Configure the phones to use persistent MTLS

Copy the following certificate related files from the openssl (Enterprise CA) to the phones' Configuration Management path (the same place as where the aastra.cfg is stored). When following chapter 19.5, the path would be to /atHome.
ca.pem - public CA signing phonecert.pem
phonecert.pem - signed client certificate
private/phonekey.pem - client private key

aastra.cfg configuration

```
sips persistent tls:1
sip outbound support:1
sip transport protocol:4 #UDP(1),TCP(2),SIP&UDP(0),TLS(4)

sips trusted certificates:"ca.pem"
sips root and intermediate certificates:"ca.pem"
sips local certificate:"phonecert.pem"
sips private key:"phonekey.pem"

sip outbound proxy:193.10.10.10
sip outbound proxy port:5061
sip srtp mode:2 #0(SRTP disabled),1(SRTP preferred),2(SRTP
only)

dynamic sip:1
sip proxy ip:0.0.0.0
sip proxy port:0
sip registrar ip:0.0.0.0
sip registrar port:0

##start: HTTPS is no different than just using persistent TLS.

https client method:"TLS 1.0"
https user certificates: "ca.pem"

action uri
startup:"https://193.10.10.10:22223/Startup?user=$$SIPUSER-
NAME$$"

###end: HTTPS
```